HELSINKI UNIVERSITY OF TECHNOLOGY
DEPARTMENT OF ELECTRICAL AND COMMUNICATIONS ENGINEERING

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# COMBINING CIRCUIT AND PACKET BASED SERVICES IN CONVERGING NETWORKS

Master's Thesis left for inspection in Espoo 13.3.2006

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**HELSINKI UNIVERSITY OF TECHNOLOGY** **ABSTRACT OF THE MASTER'S THESIS**



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The aim of this thesis is to study the functionality and the current status of the 3GPP CSICS phase 1 specifications. This thesis also evaluates how the conferencing service could be implemented with the CSICS phase 1 functionality.

First the standardization procedure of the 3GPP, the development of the mobile network and some important bodies involved in the standardization of the mobile network are introduced. Thereafter the supplementary services that have an effect on the CSICS functionality and the IMS are presented. Next the current specifications of 3GPP regarding the CSICS are studied. Last the different approaches of implementing a conference service based on these specifications are evaluated. The evaluation is based on a literature study.

As a result, the functionality described in the CSICS phase 1 specifications and evaluation of the state of the the standardization of the CSICS phase 1 are presented. Also the conference service is determined to be feasible to implement utilising the CSICS phase 1 functionality. From the recognised approaches the most suitable ones for implementation are chosen.




| | |
|---|---|
| **Keywords:** | Combinational services, CSICS, IMS, enriched call, UUS, conference service |

**TEKNILLINEN KORKEAKOULU** **Diplomityön tiivistelmä**


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Tämän diplomityön tavoitteena on tutkia sekä 3GPP:n standardoiman CSICS palvelun ensimmäisen vaiheen toiminnallisuutta että standardointiprosessin tilaa. Diplomityössä arvioidaan miten konferenssipalvelu voitaisiin toteuttaa käyttäen ensimmäisen vaiheen CSICS palvelua hyväksi.

Työn alussa esitellään 3GPP:n standardointi prosessi, matkaviestinverkon kehitys sekä matkaviestinverkon kannalta tärkeitä standardointi organisaatioita. Tämän jälkeen esitellään ne lisäpalvelut, jotka vaikuttavat CSICS:n toimintaan sekä IMS-palveluympäristö. Seuraavaksi käydään läpi julkaistujen 3GPP:n standardien mukainen ensimmäisen vaiheen CSICS-toiminnallisuus. Lopuksi esitellään tunnistetut vaihtoehdot CSICS:ää hyödyntävän konferenssipalvelun toteutukseksi sekä vertaillaan esitettyjä vaihtoehtoja toisiinsa. Arviointi on suoritettu kirjallisuustutkimuksena.

Tämän diplomityön tuloksena on kuvaus ensimmäisen vaiheen CSICS palvelun toiminnasta sekä arvio 3GPP CSICS:n standardoinnin tämänhetkisestä tilasta. Myös ensimmäisen vaiheen CSICS palvelua hyödyntävän konferenssipalvelun toteuttaminen on päätelty mahdolliseksi ja tunnistetuista toteutusvaihtoehdoista on valittu toteutettavaksi soveltuvimmat.



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|---|---|
| **Avainsanat:** | Yhdistetyt palvelut, CSICS, IMS, UUS, konferenssi palvelu |

## Foreword

This Master's thesis was written at TeliaSonera Finland, in the Mobile System Planning department. I would like to thank my superior Frej-Erik Holmström for the opportunity of doing this work.

Special thanks are due to Ville Helenius for instructing this thesis and Professor Raimo Kantola for his supervision to this thesis. I would like to thank Frans Tuomela, Jussi Laukkanen, Tommi Joutjärvi and Petri Nyberg for all the valuable help they provided.

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Sauli Österman

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## Acronyms

| | |
|---|---|
| **3GPP** | 3rd Generation Partnership Project |
| **ARIB** | Association with Radio Industries and Businesses |
| **AS** | Application Server |
| **ATIS** | Alliance for Telecommunications Industry Solutions |
| **AuC** | Authentication Centre |
| **AVP** | Attribute Value Pair |
| **BAIC** | Barring of All Incoming Calls |
| **BAIC Roam** | Barring of Incoming Calls when Roaming Outside the Home PLMN Country |
| **BAOC** | Barring of All Outgoing Calls |
| **BGCF** | Breakout Gateway Control Function |
| **BOIC** | Barring of Outgoing International Calls |
| **BOIC exHC** | BOIC except those directed to the Home PLMN Country |
| **BSC** | Base Station Controller |
| **BSS** | Base Station System |
| **BTS** | Base Stations |
| **CAMEL** | Customised Applications for Mobile network Enhanced Logic |
| **CCSA** | China Communications Standards Association |
| **CD** | Call Deflection |
| **CFB** | Call forwarding on mobile subscriber busy |
| **CFNRc** | Call forwarding on mobile subscriber not reachable |
| **CFNRy** | Call forwarding on no reply |
| **CFU** | Call Forwarding Unconditional |
| **CLI** | Caller Line ID |
| **CLIP** | Calling Line Identification Presentation |
| **CLIR** | Calling Line Identification Restriction |
| **CN** | Core network |
| **COLP** | Connected Line Identification Presentation |
| **COLR** | Connected Line Identification Restriction |
| **CRF** | Charging Rules Function |
| **CS** | Circuit Switched |
| **CSCF** | Call Session Control Function |

| | |
|---|---|
| **CSE** | CAMEL Service Environment |
| **CSICS** | Circuit Switched IMS Combinational Services |
| **CS-MGW** | Circuit Switched Media Gateway |
| **CW** | Call Waiting |
| **DTM** | Dual Transfer Mode |
| **DTMF** | Dual Tone Multi Frequency |
| **EIR** | Equipment Identity Register |
| **ETSI** | European Telecommunications Standard Institute |
| **GERAN** | GSM/EDGE Radio Access Network |
| **GGSN** | Gateway GPRS Support Node |
| **GMSC** | Gateway MSC |
| **GPRS** | General Packet Radio Service |
| **gprsSSF** | GPRS Service Switching Function |
| **GSM** | Global System for Mobile communications |
| **gsmSCF** | GSM Service Control Function |
| **gsmSRF** | Specialised Resource Function |
| **gsmSSF** | GSM Service Switching Function |
| **HLR** | Home Location Register |
| **HSDPA** | High Speed Downlink Packet Access |
| **HSS** | Home Subscriber Server |
| **HTTP** | Hypertext Transfer Protocol |
| **I-CSCF** | Interrogating CSCF |
| **IESG** | Internet Engineering Steering Group |
| **IETF** | Internet Engineering Task Force |
| **IMEI** | International Mobile Equipment Identity |
| **IMS** | IP Multimedia Subsystem |
| **IMS-MGW** | IP Multimedia Subsystem - Media Gateway Function |
| **IM-SSF** | IP Multimedia Service Switching Function |
| **MGCF** | Media Gateway Control Function |
| **MPTY** | Multi Party Service |
| **MRF** | Media Resource Function |
| **MRFC** | Media Resource Control Function |
| **MRFP** | Media Resource Function Processor |

| | |
|---|---|
| **MSC** | Mobile-services Switching Centre |
| **MSISDN** | Mobile Subscriber ISDN Number |
| **MSRP** | Message Sessions Relay Protocol |
| **MSS** | MSC Server |
| **NGN** | Next Generation Network |
| **NMT** | Nordic Mobile Telephony |
| **OMA** | Open Mobile Alliance |
| **OSA** | Open Service Access |
| **OSS** | Operator Specific Service |
| **P-CSCF** | Proxy-CSCF |
| **PDF** | Policy Decision Function |
| **PGC** | Project Co-ordination Group |
| **PLMN** | Public Land Mobile Network |
| **PS** | Packet Switched |
| **PSTN** | Public Switched Telephone Network |
| **QoS** | Quality of Service |
| **RFC** | Request For Comments |
| **RNC** | Radio Network Controller |
| **RNS** | Radio Network System |
| **RTCP** | RTP Control Protocol |
| **RTP** | Real-time Transport Protocol |
| **S-CSCF** | Serving CSCF |
| **SDP** | Session Description Protocol |
| **SGSN** | Serving GPRS Support Node |
| **SGW** | Signalling Gateway Function |
| **SigComp** | Signalling Compression |
| **SIP** | Session Initiation Protocol |
| **SLF** | Server Locator Function |
| **SMS** | Short Message Service |
| **TCP** | Transmission Control Protocol |
| **TR** | Technical Reports |
| **TS** | Technical Specifications |
| **TSG** | Technical Specifications Groups |

| | |
|---|---|
| **TTA** | Telecommunication Technology Association |
| **TTC** | Telecommunication Technology Committee |
| **UA** | User Agent |
| **UAC** | User Agent Client |
| **UAS** | User Agent Server |
| **UMTS** | Universal Mobile Telecommunications System |
| **UPD** | User Datagram Protocol |
| **URI** | Uniform Resource Identifier |
| **UTRAN** | UMTS Terrestrial Radio Access Network |
| **UUS** | User to User Signalling |
| **UUS1** | User to User Signalling Service 1 |
| **WG** | Working Groups |
| **WLAN** | Wireless Local-Area Networks |
| **VLR** | Visitor Location Register |

# 1. Introduction

## 1.1. Motivation

IP Multimedia Subsystem (IMS) is a 3rd Generation Partnership Project (3GPP) standardised network architecture that provides an access network independent standardised interface for creating services, charging mechanisms and better Quality of Service (QoS) than best effort [1]. Examples of the services that are implemented on IMS at the moment are PSTN functionality, push-to-talk, presence, instant messaging and video sharing. The big question with IMS is why it should be used. Most of the currently available IMS services could be implemented with more simple solutions, with most probably lower development costs. To be commercially successful a service utilising the IMS should have some advantageous features compared to the competing solutions accessible via the public Internet. Therefore, it is in the best interest of the manufacturers and the operators to research these types of advantages and create applications utilising the discovered benefits. The purpose of this thesis is to investigate one of these advantages; the capability to offer simultaneous circuit switched (CS) call and packed based data connection. The circuit switched CS call in UMTS Terrestrial Radio Access Network (UTRAN) or in GSM/EDGE Radio Access Network (GERAN) guarantees very good Quality of Service with a large coverage area. The packet based data transfer service in turn provides great variety with transferred media, but the Quality of Service with the real-time applications is a problem, especially when utilising GERAN [2].

## 1.2. The problem

The lack of support for the packet based real-time services via GERAN is the main motivation for this thesis. This leads to the need to study the option of creating an enriched call service in which the real-time voice media is transferred as a circuit switched voice call and other types of media streams utilise IMS packed based services. These types of services are called the Circuit Switched and IMS Combinational Services (CSICS). The standardization process in the 3GPP for these types of services is ongoing at the moment. Also the scope of currently available specifications is quite limited containing only the one-to-one types of combinational services [3]. At the moment it is interesting from the operator point of view to study the current status of the CSICS standardisation and the possibilities to offer these types of services among several subscribers.

## 1.3. Objectives, scope and methodology

This thesis has two objectives. The first is to achieve an understanding of the features included in the first phase of standardised CSICS and the status of the standardization process at the moment. Also the known problems and the recognised threats are to be studied. The second objective is to study how the standardised CSICS could be utilised to provide sessions between multiple participants. The possible approaches to provide this type of service are also evaluated.

In this thesis the focus is on functionality in the IMS domain. The non-IMS functionality is studied only when necessary and with the level of detail limited to what is required for understanding the subject. The mechanisms based on the IMS functionality on the other hand are studied in more detail. In the cases when the standardisation process is not finished, only the already finished part is utilised. The specifications are utilised in this manner even when it has been decided in the 3GPP meetings that change is to be made to the next version of the specification. These types of changes however may be described in some comments.

The thesis is done as a literature study and no measurements are therefore included in to the scope of the thesis. The study is mainly based on the 3GPP specifications and meeting reports, but also utilises technical books and the relevant standards of other standardization bodies.

## 1.4. Outline of the thesis

In the second chapter the standardization process in the 3GPP is introduced with the mobile network evolution in the 3GPP releases. The other important standardization bodies for the subject of this thesis are also presented. The purpose of the chapter is to give a general picture about the operation of the standardization process, introduce the parties involved in the developing of the mobile network and introduce the functionality and the development of the entities of the mobile network.

In the third chapter the circuit switched services specified in the 3GPP release 6 are presented. The presentation mainly contains only the services that are of significant importance to the subject of the thesis. The other services specified in the 3GPP release 6 are left out for the purpose to save space. In addition to these, the service creation environment Customised Applications for Mobile network Enhanced Logic (CAMEL) is introduced briefly.

The fourth chapter is a presentation of the IMS. First the general concepts are introduced. Then the architecture and the protocols utilised in the IMS are explained. Last the functionality of the events important for the scope of this subject is presented.

The fifth chapter describes the currently standardised CSICS. First the combinational services concept is introduced and second the motivation for these types of services is explained. Next the architecture and functionality of the standardised solution are explained. Last the known issues with this type of services are presented and the future is evaluated based on the documents published by the 3GPP.

The sixth chapter studies a conferencing service which utilises the CSICS. The functionality of the possible approaches recognised to implement this type of service is studied. The solutions developed are then compared with each other in order to find out the benefits of each solution. The last chapter, the chapter 7, summarizes the thesis and suggests subjects for further research work.

# 2. Public Land Mobile Network (PLMN) standardization

*This chapter describes briefly the history of the Public Land Mobile Network standardization and the standardization bodies involved with the IMS standardization. It also presents a short introduction to how the mobile networks have evolved.*

## 2.1. History

History of the wireless communication started in large scale at 1980s when the first generation of mobile networks was built. The Nordic Mobile Telephony (NMT) network, developed by the Nordic telecom authorities with some equipment manufacturers was one of the most important ones. This network offered the user an analogue speech service and supported the movement of the subscriber with capability to perform handovers. The Global System for Mobile communications (GSM) was developed to be the successor of NMT by European Telecommunications Standard Institute (ETSI). The GSM was labelled the second generation mobile network and utilised digital technique. GSM offered to the user not only a speech service but a text based messaging service called the Short Message Service (SMS), circuit switched data services with different capacities and the identification of the calling subscriber [4]. The GSM was very popular and it rapidly became available in almost everywhere in the world. The system has been improved during the years with the addition of a packet based data service called General Packet Radio Service (GPRS). The problem of the GSM is however the moderate capacity on air-interface which is especially a problem in large cities with a high density of population. From the need of the extra capacity was developed both the improvement to the GSM air-interface capacity called the Enhanced Data rates for GSM Evolution (EDGE) and the Universal Mobile Telecommunications System (UMTS). The UMTS was also categorised as being the third generation mobile network. The UMTS and the GSM/EDGE were planned from the start to work together with using a common core network instead of being two totally separate systems.

## 2.2. 3GPP standardization

Based on the agreement made in 1998 the 3rd Generation Partnership Project (3GPP) was founded to be the standardization body responsible for the third generation mobile network standardization. The 3GPP has no actual recognized status as being the authority that defines the standards which are to be used. It is rather a collaboration of several organizational partners. These partners are the Association with Radio Industries and Businesses (ARIB), Alliance for Telecommunications Industry Solutions (ATIS), China Communications Standards Association (CCSA), European Telecommunications Standard Institute (ETSI), Telecommunication Technology Association (TTA) and Telecommunication Technology Committee (TTC). The organizational partners have a recognised national, regional or other type of status as being the authority that defines the subjects contained to the 3GPP scope under their jurisdictions. These organizational partners form the Project Co-ordination Group (PGC) which is the group that manages the work of 3GPP [5].

In the 3GPP the technical work is done in the Technical Specifications Groups (TSG). The TSG's are further divided into the Working Groups (WG). In the spring 2005 the number of the TSG's was reduced from 5 to 4 [5]. The new organization of 3GPP including the WG's working with IMS standardization highlighted is described in figure 1 [6].

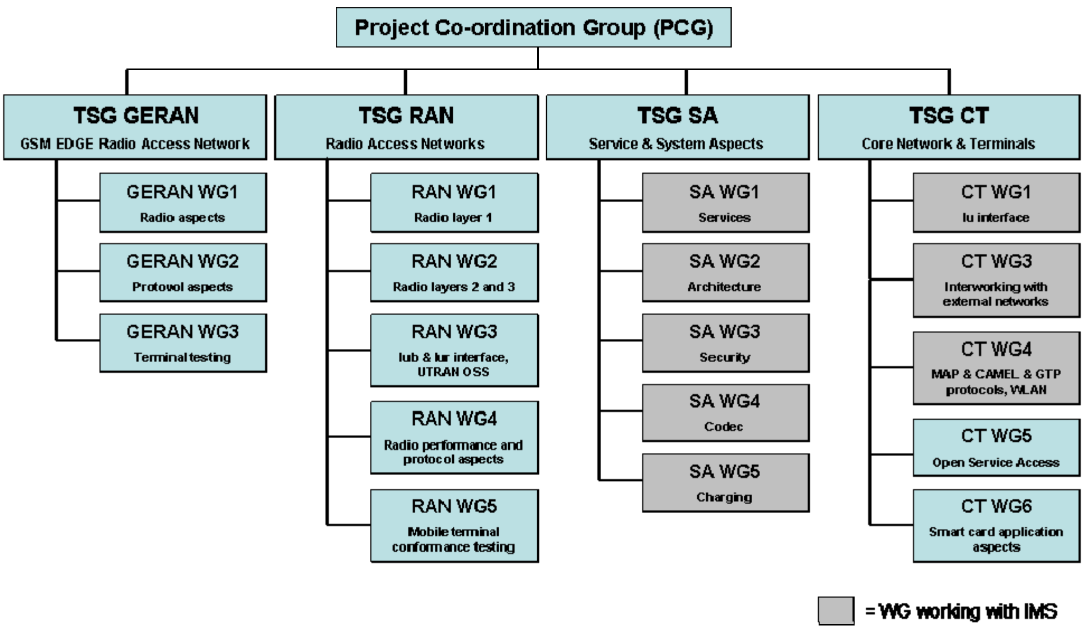


**Figure 1. The 3GPP TSG organization**

The actual work in the 3GPP will begin when at least four of the organizational members of the 3GPP are interested to study a certain feature. The feature is divided into smaller pieces which are then studied by the WG's. As a result the documents called Technical Specifications (TS) and Technical Reports (TR) are produced. The TR is a feasibility study report of a certain subject, including different views of the participants of the WG. The TS in turn is a specification that will define the recommended solution to the subject under specification. The 3GPP process for producing the TS's is divided into three stages which are explained in table 1 [5].

| Stage | Explanation |
|---|---|
| 1 | Defines service aspect of the feature from end user point of view |
| 2 | Defines logical functionality and information flows amongst the functional entities involved in providing the service |
| 3 | Specifies any necessary functionality of physical entities and the detailed protocols of the signalling between them |

**Table 1. The 3GPP defined Technical Specification stages**

The actual IMS standardization process in the 3GPP is illustrated in figure 2 [6].

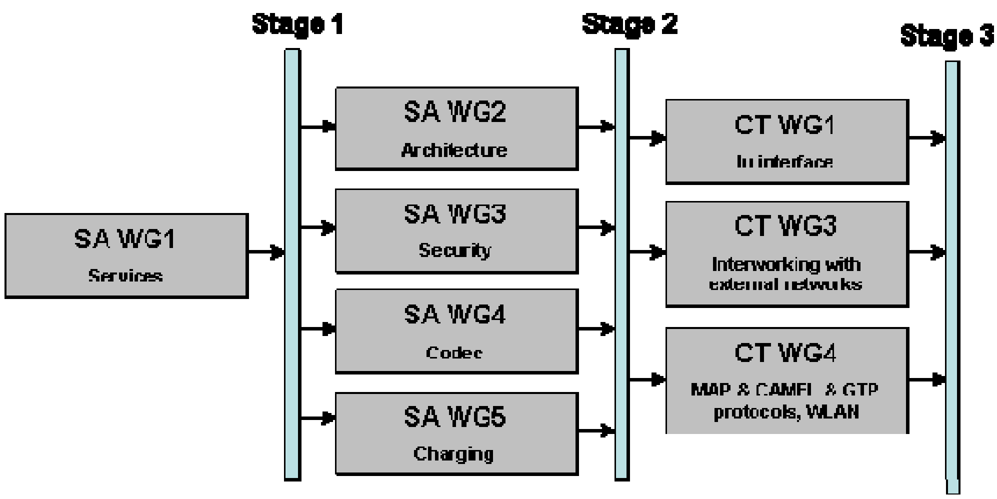


**Figure 2. The IMS standardization process**

In the naming of the TS or the TR, the first 2 digits indicate the area of the specification and the last 3 digits indicate the actual specification. The subjects of specification series are (includes 3G/GSM R99 and later) presented in table 1 [5].

This thesis concentrates on issues specified mainly in the specifications of 22, 23 or 24 series.

| Series | Subject |
|---|---|
| 21 | Requirements |
| 22 | Service aspects |
| 23 | Technical realization |
| 24 | Signalling protocols - user equipment to network |
| 25 | Radio aspects |
| 26 | CODECs |
| 27 | Data |
| 28 | Signalling protocols -(RSS-CN) |
| 29 | Signalling protocols - intra-fixed-network |
| 30 | Programme management |
| 31 | Subscriber Identity Module. |
| 32 | OAM&P and Charging |
| 33 | Security aspects |
| 34 | UE and (U)SIM test specifications |
| 35 | Security algorithms |

**Table 2. The areas of the 3GPP specifications including the 3GPP release 99 and later**

The standardization process is ongoing continuously which makes it difficult to implement applications according to specifications. For this reason the 3GPP specifications are divided into functional aggregates called releases. In practice a release is a collection of specifications that not only contain the functionality included to the previous release but also contain a substantial additional functionality. For a certain feature to be included into a release the phase one of the standardization work must be completed and phases two and three must be almost finished at the time the release is frozen. If these requirements are not met, the feature is moved to be included to the next release. The releases are frozen at a certain date, after which the specifications included in the release are no longer modified without a very good reason. The three digits in the version number of the specification indicate the release, the version and the subversion of the specification. In table 3 the 3GPP releases that are currently frozen or under development are listed [5].

| Name of the 3GPP release | Freezing date |
|---|---|
| Release 99 | March 2000 |
| Release 4 | March 2001 |
| Release 5 | March - June 2002 |
| Release 6 | December 2004 - March 2005 |
| Release 7 | date open |
| Release 8 | date open |

**Table 3*. The 3GPP releases***

## 2.3. Mobile network evolution

To able to understand how the Public Land Mobile Network (PLMN) is developing, the most important functional entities and their role in the network must be known. For this purpose the main functional entities included in the 3GPP release 99 with their functionalities and their development in the following releases are presented below.

### 2.3.1. 3GPP release 99

The Base Station System (BSS) is a radio access network of the GSM and the EDGE. It is also called in some contexts as GSM/EDGE Radio Access Network (GERAN). The BSS contains one or several Base Stations (BTS) and a Base Station Controller (BSC). The BSS hides the functionality of the radio interface from the core network and is visible to the core network only via two interfaces. The BTS is a radio modem that handles the data transport and the signalling at the radio interface. The BSC is a function that controls one or more BTS’s. The responsibility of the BTS is the radio resource allocation and performing of the handovers.

The Radio Network System (RNS) is a high capacity access network designed for the UMTS and the main difference with the 3GPP release 99 and the preceding releases. The RNS is also called in some contexts as UMTS Terrestrial Radio Access Network (UTRAN). The capacity of the radio interface of the RNS is substantially higher than the radio interface of the BSS, but the downside is that a large coverage area with RNS demands more base stations than with BSS. As the BSS the RNS also hides the functionality of the radio interface from the core network and is visible to the core network only via two interfaces. The base station of the RNS is called the Node B. The Radio Network Controller (RNC) performs similar functions in the RNS as the BSC in the BSS.

The core network Circuit Switched (CS) domain contains the functions that provide the circuit switched services. With circuit switched it is meant that the resources for the service are reserved at connection setup and released at connection release. With CS services the data is always routed via the same path during a connection. The Mobile-services Switching Centre (MSC) is an exchange that handles all the circuit switched services related switching and signalling functions. The MSC also handles the mobility of the subscriber and the data rate adaptation towards the Public Switched Telephone Networks (PSTN). The MSC can also act as a Gateway MSC (GMSC). This means that when the call is made in the network that cannot contact the Home Location Register (HLR) of the called subscriber, the call is routed to a GMSC that is capable to contact the HLR and therefore capable to route the CS call onwards. The Visitor Location Register (VLR) maintains the subscriber information of the subscribers located inside the area it is responsible for.

The core network Packet Switched (PS) domain contains the functions providing the packet based services. With the PS services the user data is encapsulated into packets which are routed towards the destination independently of each other. The Serving GPRS Support Node (SGSN) stores the subscriber data required to handle the mobile originating and the mobile terminating packet data transfer. The SGSN also handles the protocol conversion between the access network and the PS-core network and routing of the packets towards a Gateway GPRS Support Node (GGSN). The GGSN creates an interface with external IP networks. It stores the subscriber information and assigns an IP address to the subscriber.

As a final group of functions the common elements for both core network domains are presented. The Home Location Register (HLR) maintains subscriber information including allowed services for the subscriber and the VLR in which the subscriber is currently registered. The HLR provides user information to the VLR. The Equipment Identity Register (EIR) is a function responsible for storing the

International Mobile Equipment Identities (IMEIs) black listed in the network. IMEI is an identification that is unique to a certain terminal. With the EIR it is possible to prevent the usage of stolen terminals in the network. The Authentication Centre (AuC) is a function which stores the data needed for authenticating the subscriber, integrity protection and ciphering the radio path. The high level architecture of the 3GPP release 99 is described in figure 3 [7].

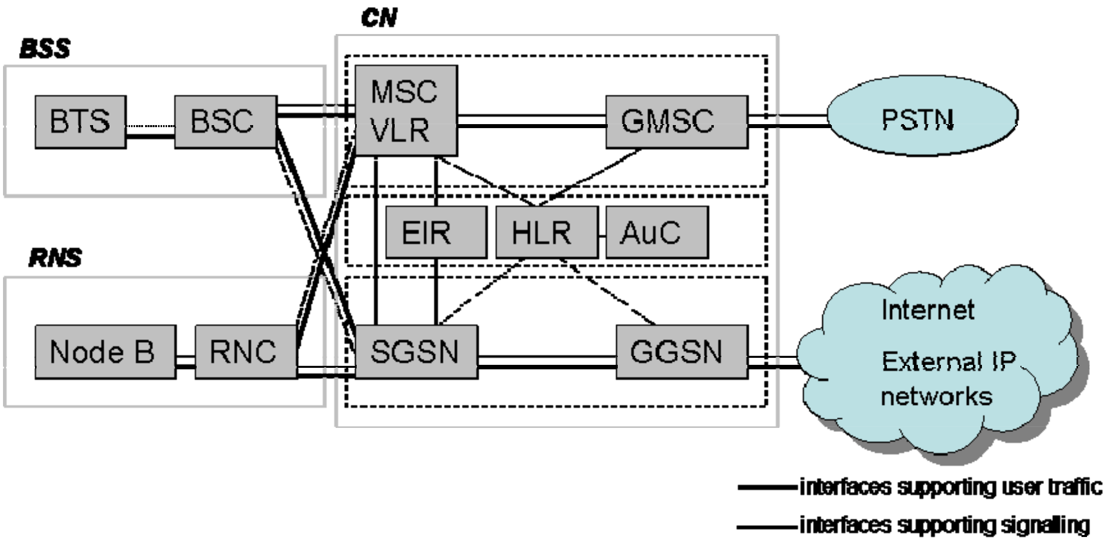


**Figure 3. The 3GPP release 99 network architecture**

### 2.3.2. 3GPP release 4

The main difference between the 3GPP release 4 and the 3GPP release 99 is the MSC functionality division into two separate parts. The MSC Server (MSS) handles the call control and the mobility portions of the 3GPP release 99 MSC functionality. The switching and interworking functionalities of the 3GPP release 99 MSC functionality are handled in the 3GPP release 4 by the Circuit Switched Media Gateway (CS-MGW). The CS-MGW is also able to use a packet based network as a transport for the media streams. The architecture of the 3GPP release 4 is described in figure 4 [8].

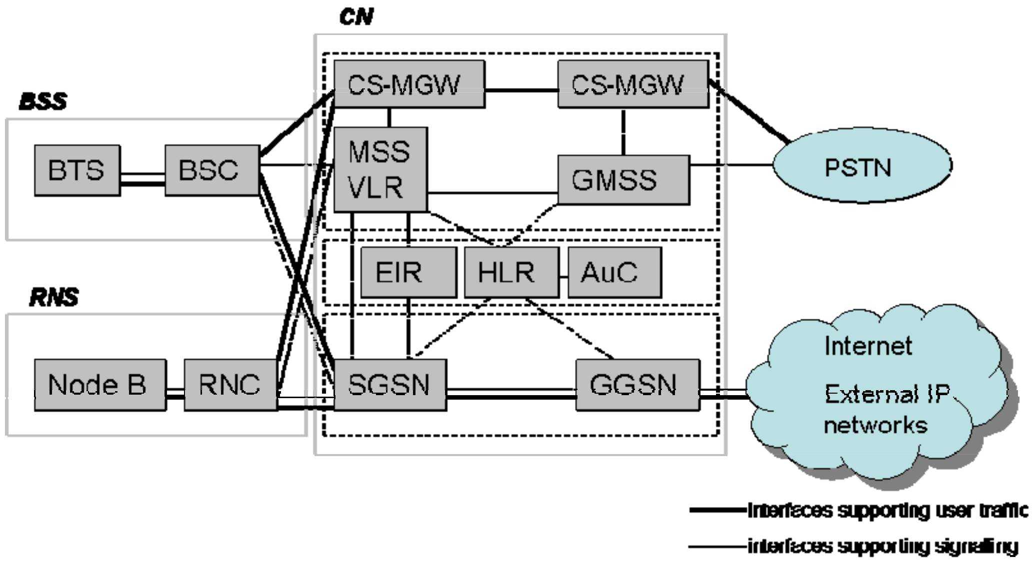


**Figure 4. The 3GPP release 4 network architecture.**

### 2.3.3. 3GPP release 5

The most important new feature included in the 3GPP release 5 is the IP Multimedia Subsystem (IMS). IMS is a standardised IP based service creation environment that offers means for audio, video or text based services utilising the PS domain as a transport. The 3GPP release 5 also introduces the Home Subscriber Server (HSS) which is a function that handles not only both the HLR and the AuC functionalities but also the storing of the IMS subscriber information and the functionality to support the control functions of the IMS. Important improvement included in the 3GPP release 5 architecture which is not visible in the architecture is that the UMTS radio network downlink capacity is considerably increased with a feature called High Speed Downlink Packet Access (HSDPA) [9]. The architecture of the 3GPP release 5 is described in the figure 5 [9].

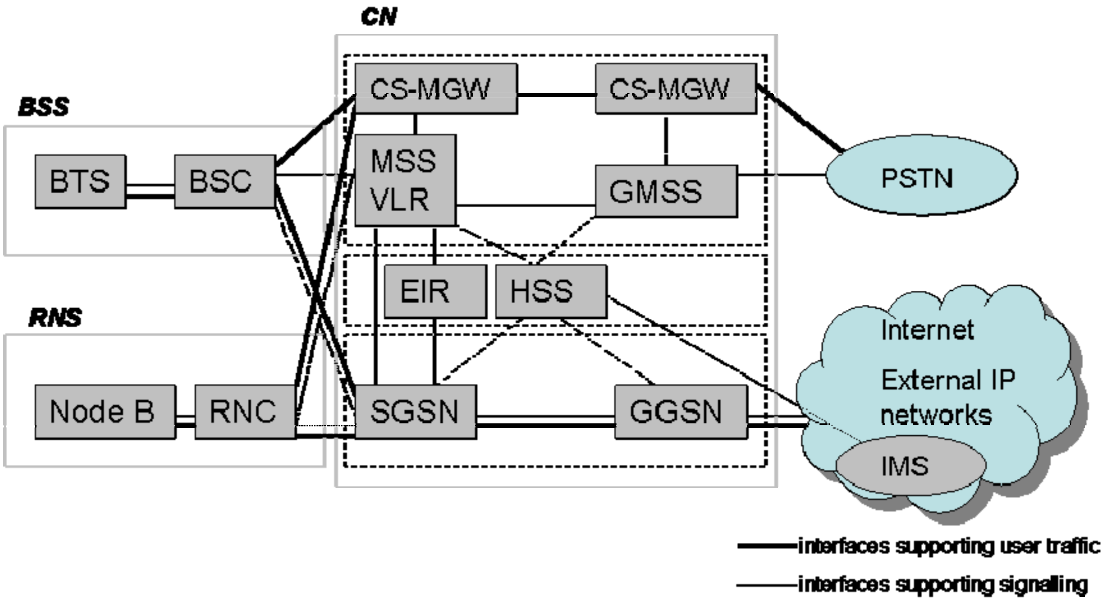


**Figure 5. The 3GPP release 5 network architecture**

### 2.3.4. 3GPP release 6

The 3GPP release 6 introduces two significant improvements. First the 3GPP release 6 introduces interworking function with Wireless Local Area Networks (WLAN). Second the policing of the IP based services is improved with the Policy Decision Function (PDF). The PDF is responsible for decisions regarding the IP bearer resource allocation. The 3GPP release 6 also introduces the Charging Rules Function (CRF) which selects and provides suitable charging rules to the GGSN. The 3GPP release 6 also includes a significant improvement of the UMTS radio network uplink capacity with a feature called enhanced uplink DCH (E-DCH) [11]. The architecture of the 3GPP release 6 is described in the figure 6 [12].

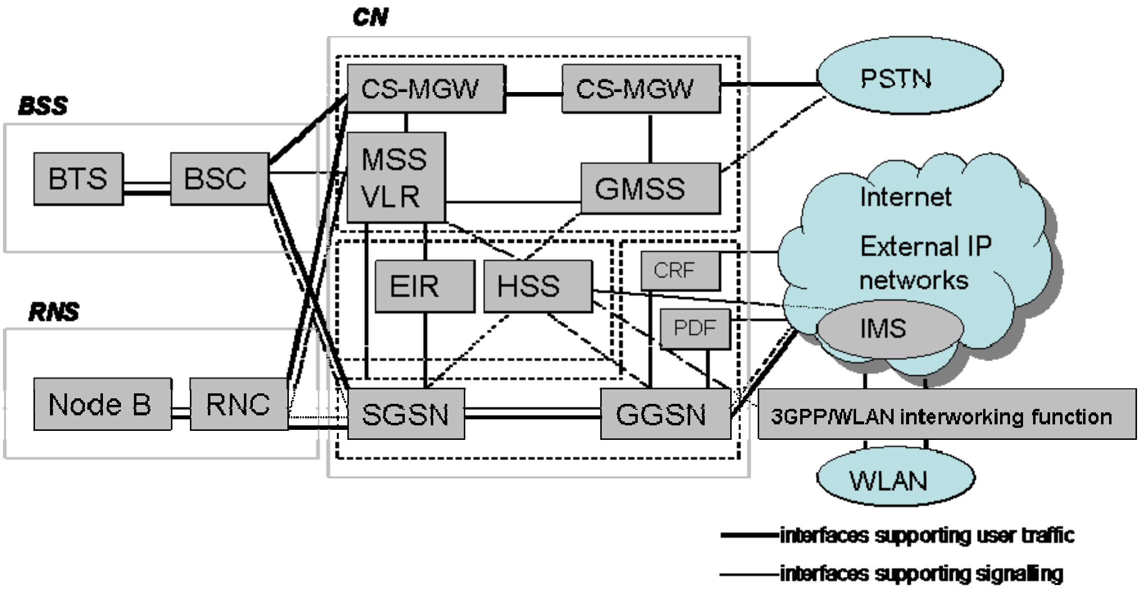


**Figure 6. The 3GPP release 6 network architecture**

The 3GPP release 6 is the latest release that has been frozen at the time of writing this thesis. In practice the most recent vendor implementations are done according to the 3GPP release 4 or the 3GPP release 5 specifications. This results that most of the operating mobile networks are a mixture of architectures presented and contain a lot of elements that are implemented according to specifications included in the 3GPP release 99 or even the earlier releases. In this document the focus is on the 3GPP releases 6 and 7. This is because the first published specifications of the combinational services are included in the 3GPP release 7 which at the time of writing this thesis is still incomplete.

## 2.4. Other important standardization bodies

Even though the actual standardization of the IMS is performed by the 3GPP; the aim of the 3GPP is to re-use existing standards of other organizations when possible. Also some of the other organizations utilise the 3GPP specification within their own work. The most important ones of these organizations are presented for the purpose of getting an overview of the parties working with the IMS.

### 2.4.1. Internet Engineering Task Force

The Internet Engineering Task Force (IETF) is an open community created for developing of the protocols for the public Internet. The open community means that any interested person can take part in the work of the IETF. The IETF will only develop specifications from subjects it has competence and interest for. All the documents produced by the IETF are public and retrievable via Internet. The actual specification work is done in the working groups. The working groups are grouped into Area Directorates that have two directors. These directors with the chairman of the IETF form the technical management team of the IETF called the Internet Engineering Steering Group (IESG). The IESG decides what IETF develops and review the finished specifications. The documents produced by the working groups are called Internet-Drafts. When a working group is finished with its work an Internet-Draft will be submitted to the IESG for reviewing. If no problems are found within the review and the IESG decides that the Internet-Draft should be published a new Request For Comments (RFC) will be published. In the IMS standardization the IETF has played a significant role as the 3GPP will re-use the IETF specified protocols when possible. Unfortunately, all the functionality required for the IMS was not provided by ready made protocols. This has been solved by forming collaboration between the 3GPP and the IETF in which the IETF defines the 3GPP required improvements and extensions to the existing protocols [13].

### 2.4.2. Open Mobile Alliance

The Open Mobile Alliance (OMA) is an organization founded for specifying the mobile services to ensure interoperability of the services. At the moment the OMA has members from more than 300 companies, consisting of operators, vendors and content providers. The services specified in the OMA specifications are not based on proprietary technologies but instead on open standards, protocols and interfaces. The applications based on the OMA standards are bearer and operating system independent and fully interoperable [13].

### 2.4.3. TISPAN

The TISPAN is a technical committee of the ETSI responsible for the standardization regarding the convergence of the fixed circuit switched and the packet-based networks. The standardization work under the TISPAN is done in the Working Groups and the Project Teams. The scope of the work of the TISPAN is the standardization of services, architecture, protocols, QoS issues, security related issues and mobility issues of fixed networks using existing technologies. The TISPAN co-operates with several other standardization bodies including the 3GPP and the IETF [15]. The future fixed network specified by the TISPAN is called the Next Generation Network (NGN). The NGN is based on the IMS specified in the 3GPP release 6 [16].

### 2.4.4. GSM Association

The GSM Association (GSMA) is a global trade association that has been founded in 1987 to promote the interest of the mobile operators. At the time of writing this thesis it consists of several hundred mobile operators and manufacturers. The objective of the GSMA is to make the wireless communication work globally. For this purpose the GSMA will produce proposals to the governments and manufacturers and perform studies for example from the area of interoperability [17].

### 2.4.5. International Telecommunication Union

International Telecommunication Union (ITU) is an international telecommunication organization within the United Nations. In the ITU the representative’s of the member countries and the commercial institutes create standards for the telecommunication area. International Telecommunication Union –Telecommunication Standardization Sector (ITU-T) is the part of the ITU that prepares the recommendations. The official objective of the ITU-T is to ensure the rapid production of the good quality standards for the telecommunications area. At the moment the ITU-T is concentrating on the standardization of the next generation networks, broadband access, multimedia services and emergency telecommunications [18].

# 3. Circuit Switched Services

*The purpose of this chapter is to introduce the circuit switched services provided by the PLMN according to the 3GPP release 6 specifications. It will not examine how these services are produced in detail, but rather presents the general picture of the circuit switched services important for the subject of this thesis.*

## 3.1. Basic and supplementary services

The circuit switched services that PLMN provides to the subscriber are divided into the basic services and the supplementary services. The basic services are further divided into the bearer services and the teleservices [19]. All the basic services are differentiated with an individual services code. The basic services include, in addition to the speech service, the circuit switched data, the short message and the fax related services. Since for the subject of this thesis these are not essential services the further evaluation is omitted. The supplementary services significant for the subject of this thesis however are explained [20].

### 3.1.1. Call Restriction Supplementary Services

The Call Restriction supplementary service group enables the prevention of the subscriber to initiate or receive specific type of calls. The service is divided into two types of restrictions, the barring of incoming calls and the barring of outgoing calls. It is possible to prevent all outgoing calls initiated by the subscriber by using the Barring of All Outgoing Calls (BAOC) supplementary service. The Barring of Outgoing International Calls (BOIC) supplementary service may be utilised if needed to prevent all outgoing international calls initiated by the subscriber. The prevention of all the outgoing international calls with the exception of the calls to the home PLMN country initiated by the subscriber is possible with activating the Barring of Outgoing International Calls except those directed to the Home PLMN Country (BOIC exHC) supplementary service. With incoming calls it is possible to deny all the incoming calls to the subscriber by activating the Barring of All Incoming Calls (BAIC) supplementary service. It is also possible to deny all the incoming calls when roaming outside the home PLMN country with Roaming

Outside the Home PLMN Country (BAIC Roam) supplementary service. The subscriber may also be provided with a password enabling the activation and the deactivation of the call barring supplementary services [21].

### 3.1.2. Call Offering Supplementary Services

The Call Offering Supplementary Services group contains four different supplementary services. These are used by the subscriber to configure the network to send all the incoming calls or the calls using specific basic service to another number if the service criterion is matched. If the subscriber wishes to forward all the incoming calls to another number, the Call Forwarding Unconditional (CFU) supplementary service is utilised. If the incoming calls are supposed to be forwarded only when the subscriber is busy, the Call forwarding on mobile subscriber busy (CFB) supplementary service is applied. If the incoming calls are to be forwarded only when the subscriber does not answer inside the specified time the Call forwarding on no reply (CFNRy) supplementary service is applied. The Call forwarding on mobile subscriber not reachable (CFNRc) supplementary service is applied when the desired functionality is that call is forwarded to another number when the called subscriber is not reachable via the network. All of these supplementary services have as optional features the possibilities that the calling party is notified when the call is forwarded and the calling party Mobile Station International ISDN Number (MSISDN) is sent to the forwarded number [22].

### 3.1.3. Call Deflection Supplementary Services

The Call Deflection (CD) supplementary service gives to the called party an opportunity to redirect an incoming call to another number before answering to the call. After the call is answered it may no longer be redirected. The CD supplementary service has a limitation of maximum five deflections per a single call. If this amount is exceeded the call is released. The CD supplementary service has as optional features a possibility that the calling party is notified when the call is deflected and the calling party MSISDN is sent to the deflected-to number [21].

### 3.1.4. Number Identification Supplementary Services

The Number Identification Supplementary Services provide the ability to present and restrict the line identity between the subscribers during the call setup or an active call. The Calling Line Identification Presentation (CLIP) is the supplementary service enabling the called party to be notified before the call is answered about the number of the calling party. The Calling Line Identification Restriction (CLIR) is the supplementary service that overrules the CLIP supplementary service and therefore prevents informing the called party about the calling party number. The Connected Line Identification Presentation (COLP) is a supplementary service enabling the call parties to be informed within the active call about the number of the other party. The Connected Line Identification Restriction (COLR) is a supplementary service overriding the COLP [24].

### 3.1.5. Call Completion Supplementary Services

The Call Completion Supplementary Services group contain two supplementary services, the Call Waiting (CW) and the Call Hold. The Call Hold supplementary service provides the means to interrupt the communication of the active call still maintaining the resources reserved for the connection reserved in case the subscriber would be willing to activate the call again. The subscriber may have only one call on hold at a time. The CW supplementary service enables the subscriber to be notified about the incoming call when engaged with another call. After receiving the notification, the subscriber has the options to accept, reject, or ignore the incoming call [25].

### 3.1.6. Call Transfer Supplementary Services

The Explicit Call Transfer (ECT) supplementary service provides to the subscriber an opportunity to connect the remote parties of two calls and release the subscribers own connections. Before utilising the ECT, the subscriber must have an active call with first of the remote parties. The call with second remote party may be either in active or in setup phase [26].

### 3.1.7. User to User Signalling Supplementary Service

The User to User Signalling (UUS) supplementary service provides the means for the transparent exchange of the subscriber generated information through the network using the SS7 signalling messages as a transport. The maximum size of the data encoded inside to a single message shall not exceed 128 octets. The UUS supplementary service is divided into three services, according to what phase the exchange of information will take place. The UUS service 1 (UUS1) enables the information exchange during the initiation and the termination of the call. The UUS service 2 (UUS2) enables the sending and the receiving of the messages after the calling party receives an indication that the called party is being alerted. The UUS service 3 (UUS3) enables the exchange of the messages during the active connection. The UUS must be provisioned to the subscriber initiating the exchange of the messages. The terminal capability to use the UUS will be verified by the network. The UUS capable terminal shall provide this information to the network without consulting the subscriber. If the UUS capability information is not provided, the terminal is interpreted to be not capable to utilise the UUS. The receiving party has no option to decide whether to accept or reject the use of the UUS. For the duration of a Multi Party Service the served subscriber may exchange messages with other participants by using the UUS3. The other participants of the Multi Party Service are not able to exchange messages between each other. An interesting issue from the interworking aspect is that some other type of networks may provide the support for the UUS but restrict the maximum length of the message to 32 octets [27].

### 3.1.8. Multi Party Supplementary Service

The Multi Party (MPTY) supplementary service enables maximum 5 subscribers to have a voice conference with each other. The MPTY supplementary service may be activated when a subscriber has two calls answered, one being in an active state and the other on hold. The subscriber who initiates the MPTY supplementary service becomes the served subscriber of the established multiParty call. During active multiParty call the served subscriber may add or remove other participants to and from the multiParty call. The other participants have only the option to remove themselves. All the subscribers participating in the multiParty call are able to put the multiParty call on hold and take part in an other simultaneous CS call. The served subscriber has a limitation to be able to take part only in one multiParty call at a time. If the served subscriber terminates the active multiParty call, the call is released for all the parties [27].

## 3.2. Customised Applications for Mobile network Enhanced Logic

The Customised Applications for Mobile network Enhanced Logic (CAMEL) is a standardized set of tools which allows a network operator to create operator specific services. The CAMEL is designed to have the capability to provide these services even when the subscriber is roaming outside the home network. The services created in the CAMEL may be triggered as a part of Mobile Originated/Terminated call, Mobile Forwarded call, Mobile Originated/Terminated IP Multimedia Session, Supplementary service invocation, Unstructured Supplementary Service Data (USSD) user interaction, Short Message Service (SMS), General Packet Radio Service (GPRS), Mobility Management events or Interrogation and control of subscription data procedures. The CAMEL service architecture is divided into several phases. The newer phase of the CAMEL includes the functionality of the previous phase with the new additional functionality. In this thesis CAMEL phase 4 functionality is studied because it is included to the 3GPP release 6.

The need to use the CAMEL based services is evaluated in the PLMN against the specified triggering criteria. If the triggering criteria is fulfilled, a request is sent to the CAMEL Service Environment (CSE). After receiving the request the CSE may activate subsequent service events to be reported to the CSE, alter information relating to the process, alter information relating to the parties involved in the process, indicate which of the possible parts of the process should occur next, perform charging activities, order in band user interaction, create additional parties in the call, release an individual call party or all call parties, connect an individual call party to the group of call parties, place an individual call party on hold or initiate a new call [29].

### 3.2.1. CAMEL architecture

The CAMEL functionality is divided into different logical functions. The GSM Service Control Function (gsmSCF) is a function that implements the service logic and control of the Operator Specific Service (OSS). The gsmSCF is always located inside the subscriber's home network. It has an interfaces with the HSS, and all the other CAMEL functions. The GSM Service Switching Function (gsmSSF) is a function that triggers the CAMEL functionality regarding the circuit switched services. The gsmSSF has interfaces with the MSS and the gsmSCF. The GPRS Service Switching Function (gprsSSF) has the same purpose as the gsmSSF, with the difference that it triggers the CAMEL functions regarding packet based services. The gprsSSF has interfaces towards the SGSN and the HSS MSS. Both gsmSSF and gprsSSF are located in the network where the subscriber is currently located. The Specialised Resource Function (gsmSRF) is a function which provides the possibility for the user interaction with the service. For example gsmSRF can play announcements to the subscribers. The gsmSRF has interfaces with the CS-MGW and the gsmSCF. The gsmSRF may be located in both the visiting and the home network. The CAMEL functions are described in figure 7 [30].

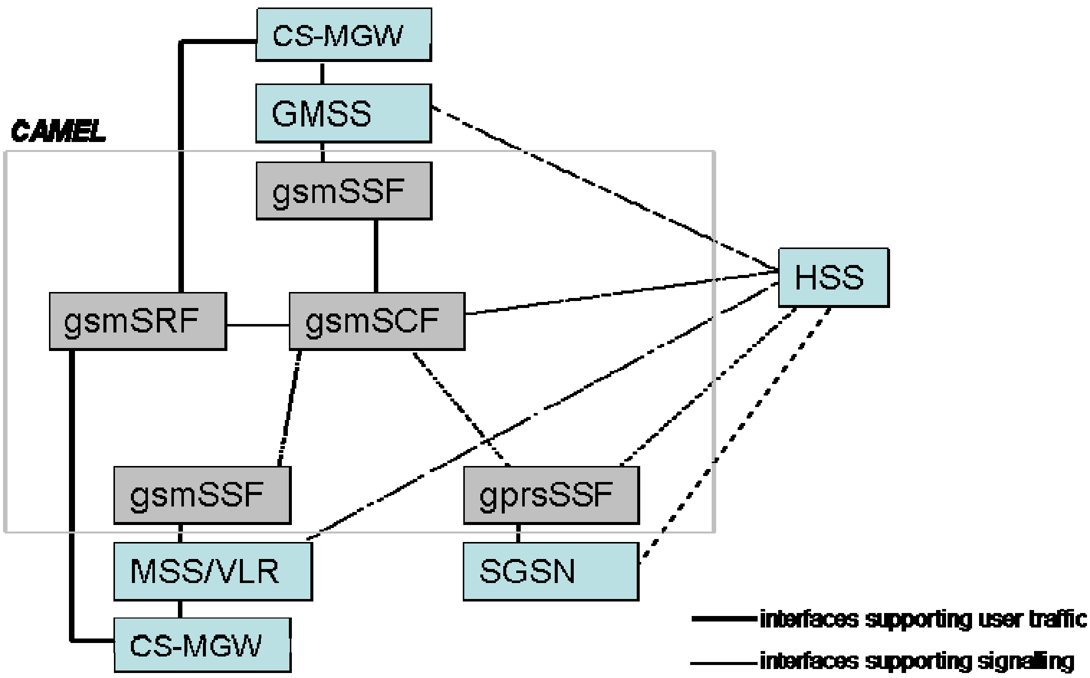


**Figure 7. The CAMEL architecture**

The interfaces between the CAMEL logical functions are not named, but instead they are called by the names of the two logical functions that they are connecting. For example the gsmSCF-gsmSRF interface.

# 4. IP Multimedia Subsystem

*This chapter will introduce IMS. The IMS architecture and the some important protocols utilised in the IMS are presented. At the end of the chapter some of the basic functionalities of the IMS are studied.*

## 4.1. Overview

The big problem of the mobile operators is that in most parts of the world the value added services created have not been very successful. The reason for this may have been that the prices have been set too high or that the services offered have just not been interesting enough. At the same time in the public Internet several successful applications have been developed. Chat or instant messaging serves as an example of these types of applications. Because the Internet is also accessible via mobile terminals and the terminal capacities have been improving all the time, the operators are facing the threat that instead of selling services they would be forced to sell only the transmission service for the services produced in the public Internet. This is the challenge the IMS was developed to meet. With the IMS the 3GPP is not standardising the actual services but the means to develop these services. If the services are standardised, they are done by the OMA or the GSMA.

The IMS is a standardised access network independent service environment that provides the connectivity between parties, means for negotiation of the Quality of Service (QoS), support for interworking with other circuit switched networks, support for interworking with other packet switched networks, support for roaming and means for charging [1].

## 4.2. User identity in IMS

In every type of network the subscribers must be identified with some method to provide the most important service, the connectivity. Also the operator may want to restrict the usage of the network in which case the subscriber must be identified before allowing the use of the network. In the IMS these needs are handled by two types of identifications of the user. First every subscriber is given at least one private user identity. This private user identity is meant to be kept secret and used during the IMS registration procedure together with other authentication information. For each private user identity there must be at least one public user identity. The public user identity is used for routing the messages between subscribers. The public user identity is public information and shared with other subscribers like the email addresses or the telephone numbers. The public user identity is attached to exactly one service profile. The service profile determines which services the subscriber is allowed to use and when these services will be triggered. The relationships between public user identities, private user identities and service profiles are described in the figure 8 [31].

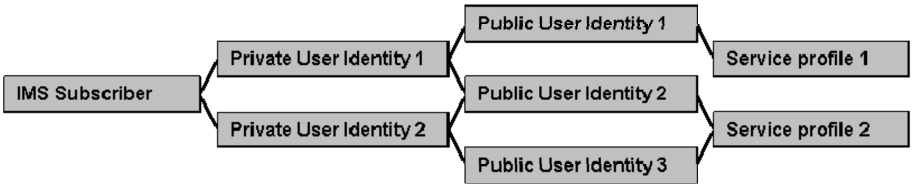


**Figure 8. The relationships between public user identities, private user identities and service profiles in the IMS**

The format of the private user identity is the network access identifier (NAI), for example username@realm. The public user identity has two possible alternative formats: Session Initiation Protocol Uniform Resource Identifier (SIP URI) and tel URI [32]. SIP URI format in its basic form is sip:username@operator.com. The tel URI format is similar to the telephone number, the tel:+358401234567 serves as an example.

## 4.3. IMS architecture

The basic idea of the IMS is that while the signalling data is always routed through the IMS logical functions the user data is routed via shortest possible path. Therefore most of the IMS logical functions are involved only with signalling traffic [1].

### 4.3.1. Call Session Control Functions

In the IMS the logical functions responsible for the call control are called Call Session Control Functions (CSCF). The IMS function which the terminal will contact is called Proxy-CSCF (P-CSCF) and it is allocated to the subscriber during the IMS registration procedure. The P-CSCF is the only contact point for the subscriber to the IMS. This means that all the signalling traffic to or from the subscriber is routed via the P-CSCF. The P-CSCF determines the Interrogating CSCF (I-CSCF) located at the home domain of the subscriber. The P-CSCF also performs signalling compression/decompression and authorises bearer resources for the subscriber. The P-CSCF can be located either in the subscriber’s home network or in the visiting network. The I-CSCF has three major tasks to perform. First it hides the topology of the subscriber's home network and second it allocates the Serving CSCF (S-CSCF) for the subscriber during the IMS registration procedure. The third task of the I-CSCF is to lookup the S-SCSF of the called subscriber in case of incoming session initiation request. In practice the I-CSCF is located in the subscriber's home network. The S-CSCF is the centre node of the IMS. The S-CSCF authenticates the subscriber by using authentication data retrieved from the HSS. After authentication of the subscriber the S-CSCF binds the user's current IP address and the subscriber identity until the subscriber is deregistered from the IMS. The S-CSCF also informs the HSS about the changes of the registration state of the subscriber. Other task of the S-CSCF is to trigger the services when required. For this purpose the S-CSCF retrieves the subscriber related filter criteria information from the HSS, which it then uses to detect if certain application is to be evoked during the IMS procedures. The S-CSCF also prevents the subscriber for performing unauthorized operations .The S-CSCF is always located in the subscriber’s home network [1].

### 4.3.2. Functions for storing subscriber related information

In the IMS the logical function that is responsible for storing the user related subscription information is the Home Subscriber Server (HSS). The HSS is a database that stores the subscriber location information, security information, subscriber’s service information and the identity of the S-CSCF allocated to the subscriber. The Server Locator Function (SLF) is a function required when the IMS network contains more than one HSS. In this case, the SLF returns when queried to the CSCF the information about which of the HSS's is containing the information of a specific subscriber [1].

### 4.3.3. Application servers

An IMS session between two subscribers does not require an application server (AS) to be involved. But the purpose of the IMS was not to offer plain connectivity service but rather to offer value-added multimedia services. The AS is a logical function in the IMS that implements these types of services. It may be located in the subscriber's home network or in the third party network. There are three types of application servers in the IMS. These are: the Session Initiation Protocol Application Server (SIP AS), Open Service Access Service Capability Server (OSA-SCS) and CAMEL IP Multimedia Service Switching Function (IM-SSF) [3].

The SIP AS is the host for the native type of SIP services in the IMS utilised to produce the required multimedia services for the subscribers. The Open Service Access (OSA) is a 3GPP specified architecture that enables the developers of the services to use the IMS network functionality through a standardized interface [32]. The OSA-SCS provides the interface for the OSA applications to be able to utilise the IMS network functionality. Using the OSA-SCS is a secure way for the operators to offer the possibility to the third parties to offer their services via the IMS. The third type of IMS AS 's is the IM-SSF- The IM-SSF provides the means for utilising the CAMEL services through IMS and vice versa. The functionality of the IM-SSF is similar to the gsmSSF in the CAMEL architecture. All three types of AS’s look exactly the same to the S-CSCF’s. An interface with the HSS and the AS is possible only if the AS is located in the home network [1].

### 4.3.4. Media Resource Function

The Media Resource Function (MRF) is a logical function that may send and receive user plane data streams in the IMS. This functionality may be utilised to create conferencing services, to perform transcoding functions or to play announcements. The MRF is further divided into two parts. The Media Resource Control Function (MRFC) is responsible for the functions related to signalling and for the controlling of the Media Resource Function Processor (MRFP). The MRFP is responsible for the user plane functionality [1].

### 4.3.5. Functions providing interworking with PSTN and PLMN

The IMS is designed to be able to communicate with the PLMN CS domain and the PSTN networks. To provide this type of functionality the IMS uses four logical functions. These are: the Breakout Gateway Control Function (BGCF), the Signalling Gateway Function (SGW), the Media Gateway Control Function (MGCF) and the IP Multimedia Subsystem - Media Gateway Function (IMS-MGW). The BGFC is responsible for the selection of the IMS network in which the breakout to the circuit switched network will happen. If the breakout will happen in the home IMS network, the BGCF will select the MGCF to handle the session further. The MGCF performs the high level signalling protocol conversion and the SGW performs the low level signalling conversion required for communication between the IMS and the circuit switched networks. The MGCF is also responsible for controlling the IMS-MGW. The IMS-MGW connects the media plane of the IMS to the media plane of the circuit switched networks and if necessary performs transcoding of the media streams. The figure 9 describes the architecture of the IMS [1].

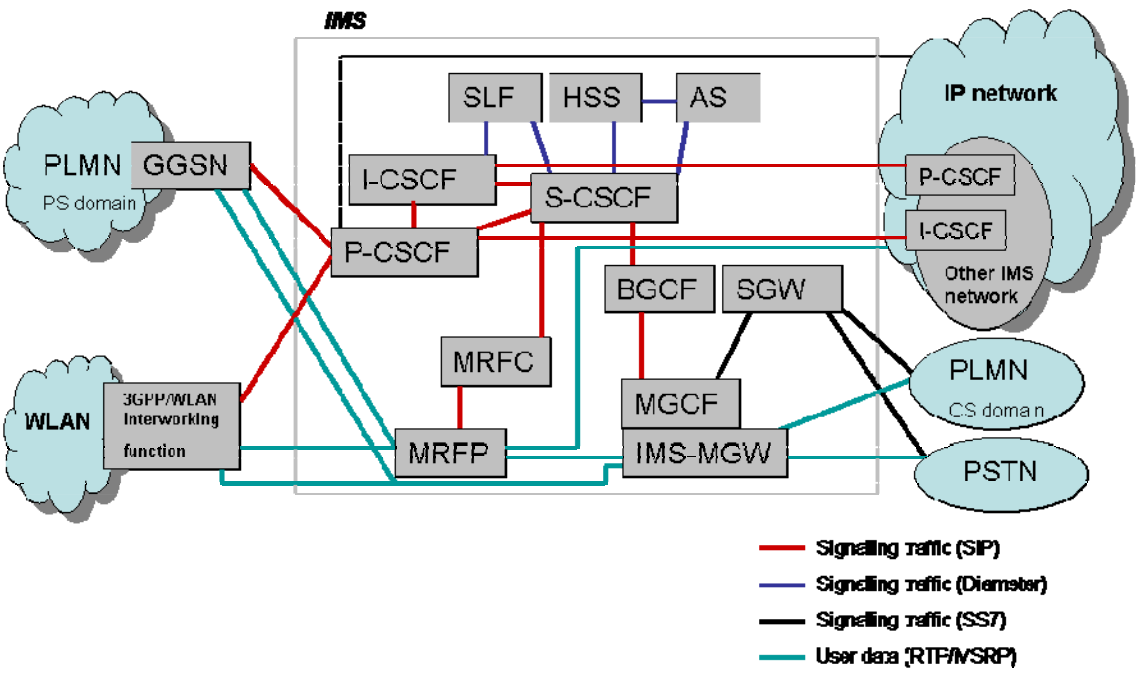


**Figure 9. The IP Multimedia Subsystem architecture**

## 4.4. Protocols in the IMS

The IMS uses a wide variety of protocols to perform different tasks. In this thesis only some of the most important protocols will be introduced. The presented application level protocols have been collected into figure 10.

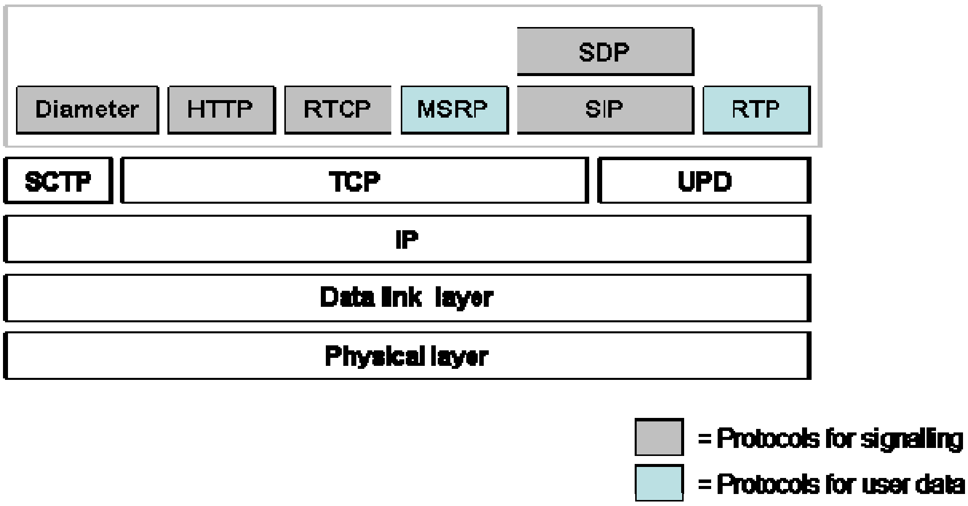


**Figure 10. Summary of the protocols used in the IMS**

### 4.4.1. Session Initiation Protocol

The Session Initiation Protocol (SIP) was originally developed by the IETF for the multicast purposes but it was selected with additional extensions to be also the protocol that performs the session control in the IMS. SIP is a text based protocol which makes the troubleshooting easier but on the other hand also increases the amount of traffic load caused by signalling when compared to the signalling protocols that use a binary format. The amount of the SIP signalling traffic is a problem for the low bandwidth links. For this purpose the IETF has specified the Signalling Compression (SigComp) as an optional feature [1].

SIP is a control protocol that may be used for establishing, modifying, and terminating multimedia sessions. With SIP it is possible to invite new participants to the existing sessions and modify the media of the existing session. SIP defines the following entities: User Agent Client (UAC), User Agent Server (UAS), registrar, redirect server and proxy server. The UAC and the UAS are the endpoints of SIP signalling. The registrar stores the location information of the user. The Proxy servers route the message toward the destination and the redirect server informs the sending party about the destination the message should have been sent to. SIP works with SIP dialogs between UAC and UAS. One SIP dialog may contain several transactions, which in turn contain one SIP request and usually one or more SIP responses. The responses that do not end the transaction are called to provisional responses [1].

A SIP message consists of the start line, header fields and the body. The start line of a SIP request contains a SIP method name, SIP URI of the receiver and version of the protocol. Since SIP has been designed to be extensible, new methods may be added to the protocol without affecting the functionality of earlier implementations. The six SIP methods included in the core protocol are introduced in the table 4 [32].

| Method | Functionality |
|---|---|
| ACK | Acknowledges that a session is established |
| BYE | Terminates a session |
| CANCEL | Cancels request that is yet to be accepted |
| INVITE | Establishes a session |
| OPTIONS | Queries the capabilities of a server |
| REGISTER | Registers an SIP URI to the IMS |

**Table 4. The methods of the core SIP protocol**

The SIP methods specified in the protocol extensions and used in the IMS are introduced in the table 5 [1].

| Method | Functionality |
|---|---|
| NOTIFY | Notifies about a particular event |
| PRACK | Acknowledges that a provisional response has been received |
| PUBLISH | Updates a server with new information |
| SUBSCRIBE | Requests that sender is notified when specified events occur |
| UPDATE | Modifies ongoing session |
| MESSAGE | Transports a text based message |
| REFER | Requests a server to send a request |
| INFO | Used as a transport for PSTN signalling |

**Table 5. The additional SIP methods used in the IMS**

In the SIP response the start line contains the protocol version and the status of the request. The status of the request is presented both in a numerical format and in a textual format. The SIP responses are divided into classes based on their numerical codes. The SIP response classes with their general meanings are presented in table 6 [32].

| Number code range | Class name | Explanation |
|---|---|---|
| 100-199 | Provisional | Request received, continuing to process the request |
| 200-299 | Success | The action was successfully received, understood, and accepted |
| 300-399 | Redirection | Further action needs to be taken in order to complete the request |
| 400-499 | Client Error | The request contains bad syntax or cannot be fulfilled at this server |
| 500-599 | Server Error | The server failed to fulfil an apparently valid request |
| 600-699 | Global Failure | The request cannot be fulfilled at any server |

**Table 6. The SIP response classes**

A SIP header field consists of the header name, a colon and the header field’s value. Some of the SIP header fields are mandatory which means that every SIP message will contain those. Also some of the header fields may have multiple entries in the same SIP message. Some important SIP header fields are collected in the table 7 [1].

| Header field | Content | Used for |
|---|---|---|
| To (mandatory) | SIP URI of the of the receiver of the message | Filtering or human consumption, not routing |
| From (mandatory) | SIP URI of the sender of the message | Filtering or human consumption, not routing |
| Cseq: (mandatory) | Sequence number and SIP method name | Matching requests and responses in SIP transactions |
| Call-ID (mandatory) | Unique identifier for the SIP session | Identifying to which SIP session message belongs to |
| Max-Forwards (mandatory) | Number of times the SIP message may be forwarded | To prevent eternal routing loops |
| Via (mandatory) | Identity of all proxies via SIP request has gone trough | To ensure the reply of request will also go trough the same path as request did |
| Contact (optional) | SIP URI of the sender | To route future SIP request to the sender |
| Record-Route (optional) | Identity of all proxies via SIP request has gone trough that want to be included in to the signalling path of the future SIP messages concerning particular session | To ensure that necessary proxies stay on session signalling path |
| Route (optional) | Identity of all proxies in the received Record-Route field | Routing messages trough correct proxies |
| Supported (optional) | The SIP extensions supported by the sender that are not in the required header | Negotiations of the SIP extensions that can be used in the dialog |
| Require (optional) | The SIP extensions the sender wants to use in the dialog | Negotiations of the SIP extensions that can be used in the dialog |
| Unsupported (optional) | The SIP extensions not supported by the sender but requested by the receiver | Negotiations of the SIP extensions that can be used in the dialog |
| Alllow (optional) | SIP method names that the sender supports | Informing the receiver of what SIP methods can be used |
| Content-Type | Indication of message body content nature | Informing what message body is |
| Content-Length | Length of the message body | Informing where message body ends |

**Table 7. Some important SIP header fields**

The body of the SIP message is located after the header fields. The header fields and the message body are separated from each other with one empty line. The contents, length and appropriate handling of the message body are described in the header fields. Usually the SIP message body is a session description coded according to the Session Description Protocol (SDP) but it may be for example plain text. It is also possible for a SIP message to carry multi-part bodies [1].

### 4.4.2. Session Description Protocol

The SDP is an IETF specified text based protocol or more accurately an IETF specified text based format for describing multimedia sessions. The SDP description may be divided into three parts. The first part describes the necessary session level information including the session name and the address to contact. The second part of the SDP description describes the time level information such as when the session is to be terminated. The third part describes the media level information, for example connection information of a particular media stream. The format of the SDP is very simple; every line starts with a type letter which is then followed by an equal sign and the actual information. The lines in the SDP description must be in a specified order. The number of the type letters is left to be quite small. The flexibility for the future need is planned to be met by using the media attributes under the type letter a. If the type letter or the attribute is not known to the SDP parser it will be discarded [34].

### 4.4.3. Diameter

Diameter is an IETF specified protocol for Authentication, Authorization and Accounting (AAA). Diameter was developed to be the successor of the Remote Authentication Dial In User Service (RADIUS) protocol because RADIUS was missing the functionality required in large scale environments. Diameter is designed to be an extensible protocol and for this purpose the implementation of the protocol has been divided into two parts. The base protocol consists of the basic functionality which every Diameter node must be able to provide. The actual applications are considered to be extensions to the basic functionality. From the transport layer protocol the Diameter expects reliability and congestion control, qualities offered for example by Transmission Control Protocol (TCP). The Diameter is a peer-to-peer protocol, meaning that any one of the Diameter entities is able to send requests to any other Diameter entity. The Diameter signalling consists of sessions which may contain several individual requests and replies. A Diameter message contains a header and one or more Attribute Value Pairs (AVP). The header contains the information about the protocol version, indication of the message request or reply and the command code. The command code identifies the meaning of the request/answer. To achieve the required functionality of the IMS the 3GPP has standardised necessary extensions to the Diameter base protocol.

The actual information in the message is coded into the Attribute Value Pairs. A single message may contain several AVPs. The AVP may be mandatory or optional to the specific command code. The AVP’s contain an AVP code, Vendor-ID, AVP length and the actual data. The Vendor-ID field and the AVP code together identify the meaning of the AVP. For example the Vendor-ID of the IETF is 0 and the Vendor-ID of the 3GPP is 10415 [1].

### 4.4.4. Real-time Transport Protocol

The Real-time Transport Protocol (RTP) provides means for an unreliable end-to-end delivery service of real-time data. In addition to data transfer, RTP provides information about the type of the payload, sequence numbering, time stamping and the identity of the sender. The extendibility to all types of applications is achieved with the RTP by defining the profiles and the formats for a particular application. The profile defines the syntax and the semantics of the extension header fields. The format in turn will define how the data after the header is to be interpreted. The existence of these extensions is indicated in the RTP header field X. The RTP packets are delivered over User Datagram Protocol (UDP). The RTP does not offer means to provide the quality of service (QoS) however an indication of the achieved QoS is provided by the RTP Control Protocol (RTCP). The RTCP is a protocol that is always used together with the RTP. The RTCP packets are sent periodically between all participants of the session. The RTCP packet sending rate is adjusted with the number of participants in order to keep the amount of the signalling traffic at a reasonable level. The RTCP utilises different packet types, including sender reports, receiver reports, source descriptions and application specific control packets. The sender and the receiver reports are used for reporting the perceived quality. The source description describes the physical sender and may be utilised to combine several different RTP streams coming from the same sender. The RTP and the RTCP are both specified by the IETF [35].

### 4.4.5. Message Sessions Relay Protocol

The Message Sessions Relay Protocol (MSRP) is a text-based, connection-oriented protocol for exchanging arbitrary binary content. It is designed specially for the instant messaging applications. The MSRP sessions are negotiated as any other types of sessions within SIP signalling. The SIP offers the possibility to instant messaging with the MESSAGE method. The benefits of using the MSRP over the SIP MESSAGE method are the negotiated security, integration with other media types and direct client-to-client operation. Also the MSRP does not limit the size of the messages sent, but instead it offers means to deliver the message in smaller parts and means to reassemble the message at the receiving end of connection. The MSRP defines two methods, the SEND and the REPORT. The SEND is used to deliver the messages or the parts of the messages. The REPORT is used to report the status of the previously sent messages. The REPORT number codes follow the same pattern as with SIP replies. As the transport protocol the MSRP uses the TCP. The MSRP is an IETF specified protocol but at the moment the specification is available only as an Internet Draft [36].

### 4.4.6. Hypertext Transfer Protocol

The Hypertext Transfer Protocol (HTTP) is a text based protocol meant for the exchange of information. The HTTP is know for being utilised by the World-Wide Web (WWW) service, but it can be used to exchange any type of data between the server and the client. The HTTP uses requests and replies to provide the required functionality. The functionality and the format of the HTTP are very close to SIP, which is no wonder since HTTP was one of the protocols that were the models for the SIP. There are seven different type of possible requests defined in the HTTP version 1.1. These requests are listed in the table 8 [37].

| Operation | Description |
| --- | --- |
| GET | Retrieves any entity (file) identified by the URI |
| HEAD | Retrieves the metainformation contained in the headers of the file identified with request URI |
| POST | Request the server to accept the entity in the request as a new subordinate of the resource identified by the URI |
| PUT | Requests that the entity in the request would replace the file identified by the URI |
| DELETE | Requests that the server would delete the resource identified by the URI |
| TRACE | Used to invoke a remote, application-layer loop-back of the request message |
| CONNECT | Used for tunnelling purposes |

**Table 8. The HTTP v 1.1 requests**

As with SIP, the HTTP responses contain the number code indicating how the operation requested has succeeded. The HTTP uses otherwise the same codes as SIP with the exception of the Global Failure (600-699) class which is not included in the HTTP version 1.1.

## 4.5. Functionality

### 4.5.1. IMS registration procedure

The IMS registration is an event that binds the public user ID to the IP address from which the public user ID is reachable. Within this procedure the user is allocated with S-CSCF handling the session control for the subscriber. The process starts with the user sending a SIP REGISTER request to the P-CSCF. The P-CSCF address may be previously known to the subscriber or provided by other means. The P-CSCF will then locate the I-CSCF by using the Domain Name Service (DNS) and the domain name part in the public user ID. After resolving the address of the I-CSCF the P-CSCF will send the SIP REGISTER request towards it. The I-CSCF in turn locates the S-CSCF allocated to the user. If none exist the I-CSCF will select an appropriate S-CSCF for the subscriber. To achieve this the I-CSCF requires the support of the HSS. If the network contains more than one HSS, the I-CSCF must first resolve which of the HSS's is containing the subscriber's information. For this purpose the I-CSCF will send the same Diameter UAR request that it will eventually send to the HSS to SLF. The SLF replies to the I-CSCF with the Diameter UAA message containing the address of the correct HSS. When the I-CSCF is aware of the correct HSS it will send the Diameter UAR request to the HSS. The HSS will reply with Diameter UAA containing the address of the allocated S-CSCF or the parameters that the I-CSCF may use as a basis for the selection of the S-CSCF. Then finally the SIP REGISTER request is sent to the S-CSCF. The S-CSCF must authenticate the subscriber and for this purpose it retrieves with Diameter MAR request and the Diameter MAA reply the authentication information from HSS and denies the registration by sending the SIP 401 unauthorized reply to the subscriber. This reply will contain the challenge to which the subscriber must calculate the reply utilising the shared secret known also by the HSS.

Then the subscriber will send a new SIP REGISTER request containing the reply to the challenge presented. When the new SIP REGISTER request arrives to the S-CSCF, it will verify that the reply to the challenge is correct and retrieve the profile of the public user ID from the HSS. The S-CSCF will also inform the HSS that the public user ID is registered under it. After this the S-CSCF will send the SIP 200 OK reply to the subscriber. This procedure is described without the DNS query in the figure 11 [1].

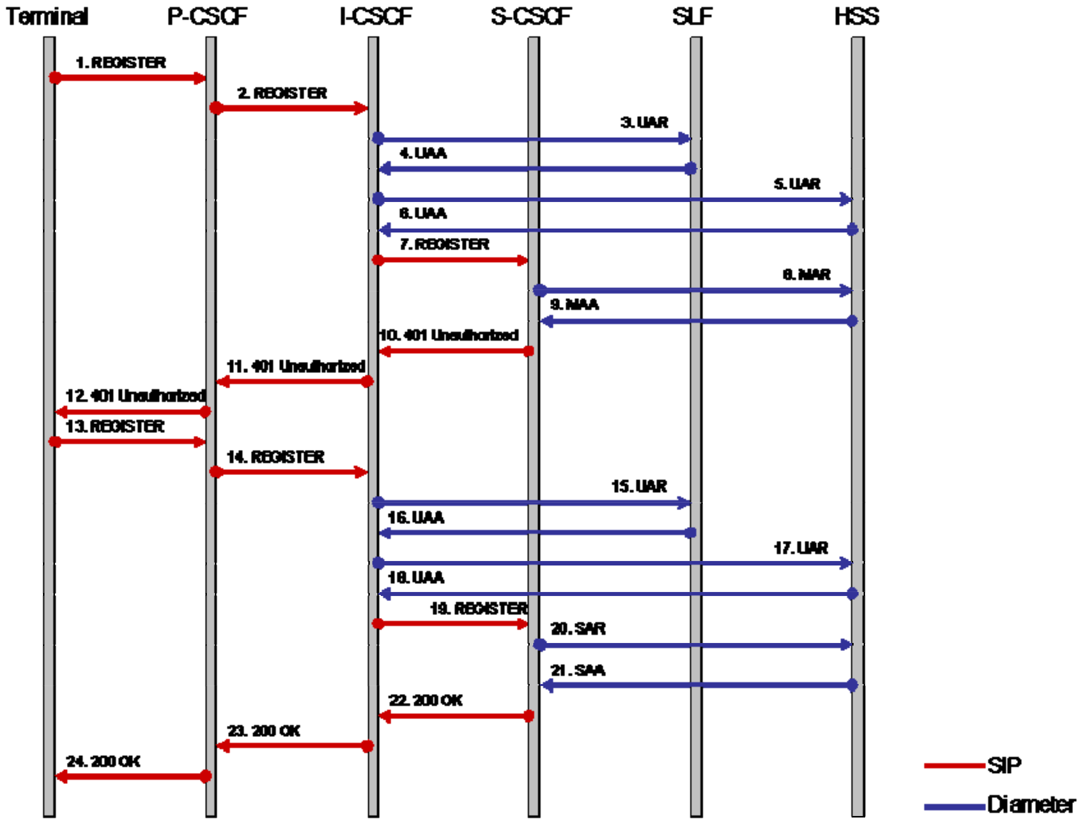


**Figure 11. The IMS registration procedure**

### 4.5.2. IMS session setup procedure

The IMS session setup is quite an extensive procedure that contains several messages sent back and forth. For the space saving purposes the recourse reservation portion has been excluded from the session setup presentation and for the same reason the presentation describes the session setup in the situation when both of the subscribers are located under same the S-CSCF. The requirements of the session setup are that both the calling party and the called party are registered in the IMS. The session setup begins when the terminal 1 sends the SIP INVITE to the terminal 2. The SIP INVITE contains the nodes allocated during registration in the route header and the session description of the preferred session. The P-CSCF returns the SIP 100 Trying as a indication to the terminal 1 that it received the SIP INVITE message. Then the P-CSCF will send the SIP INVITE to the S-CSCF which in turn replies to the P-CSCF with the SIP 100 Trying message. At this point the S-CSCF evaluates the SIP INVITE against the filtering criteria's of the subscriber. In another words the S-CSCF determines whether the SIP INVITE should be routed towards some AS instead of routing it towards the terminal 2. Because in this example both of the subscribers are under the same S-CSCF and located in the home network, the S-CSCF will send the SIP INVITE to the P-CSCF serving the terminal 2. After the receiving of the SIP INVITE the P-CSCF serving the subscriber 2 will reply to the S-CSCF with the SIP 100 Trying and send the SIP INVITE to the subscriber 2. The terminal 2 in turn indicates that it has received the message by sending the SIP 100 Trying to the P-CSCF.

This is the point when the SDP negotiation and the reservation of the resources will happen, but in this description, it is omitted. After the resources have been reserved the terminal 2 will alert the subscriber 2 and send the SIP 180 Ringing message back to the terminal 1. The terminal 1 replies with SIP PRACK message, only to indicate to the terminal 2 that it has received the SIP 180 Ringing message. After receiving the SIP PRACK message the terminal 2 will reply to the terminal 1 with the SIP 200 OK message, again only to indicate that the PRACK message has been received. What has been accomplished after all this is that the characteristics of the session have been negotiated and the terminal 2 is alerting and both of the terminals are aware of the situation. What follows is that the subscriber 2 accepts the session, which results that the SIP 200 OK is sent to the terminal 1. The terminal 1 replies to this SIP 200 OK with ACK. Now the direct transmission of the user data between the terminal 1 and the terminal 2 may begin. The IMS session setup has been described in the figure 12 [1].

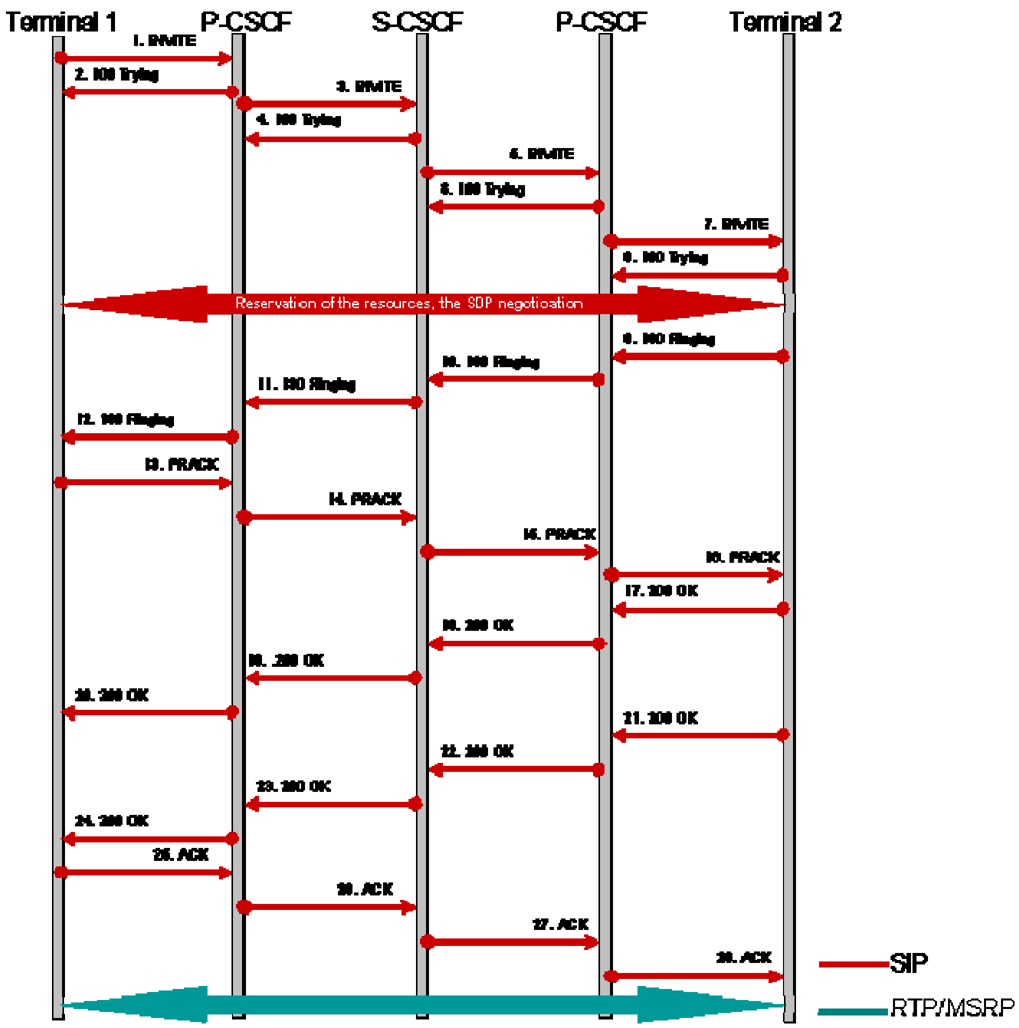


**Figure 12. The IMS session setup procedure**

### 4.5.3. AS functionality

The S-CSCF determines for every initial SIP request if the message is supposed to be routed towards an AS. With the initial SIP request is meant either a stand alone request or a request that initiates a SIP dialog. When the initial SIP request arrives to the S-CSCF, the S-CSCF compares if the request matches to any of the filtering criteria's concerning the subscriber. The filtering criteria can be for example a SIP INVITE to the specific SIP URI. If there is a match, the S-CSCF will route the message towards the specified AS.

The AS has three ways it is able to operate on. First it is able to act as a User Agent (UA), which means that it will be the other endpoint of the SIP dialog. The other option for the AS is to act as a proxy, meaning that it routes the messages forward to some other destination with perhaps changing some of the header field values inside the message. The third way the AS is able to act is a back-to-back UA. This means that the AS will terminate one session and initiate an other based on the information in the initial session request. The three roles of the AS have been described in the figure 13 [1].

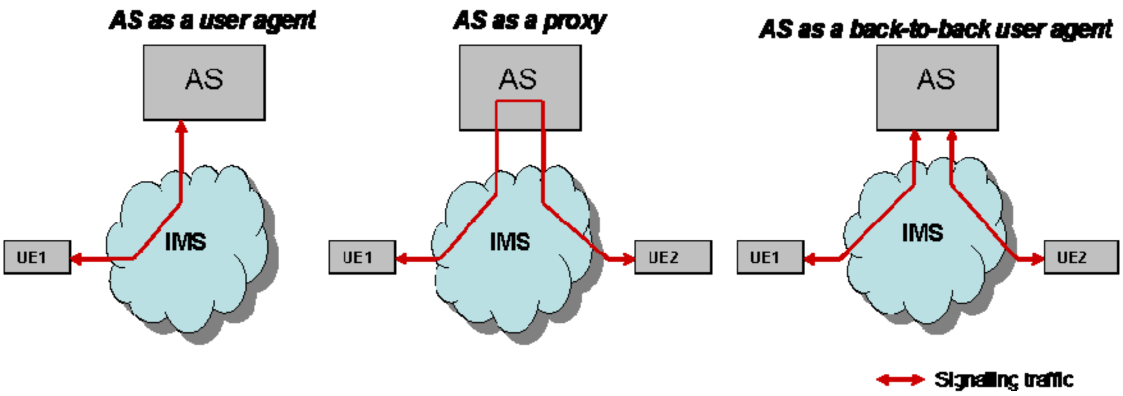


**Figure 13. The three roles of the Application Server**

### 4.5.4. MRF functionality

The MRF is a function which the AS may utilise to provide services that require exchanging of media streams with terminal equipments. When the S-CSCF receives an initial SIP request and finds a match with some filtering criteria, it will route the request to the AS indicated in the filtering criteria. The AS examines the message and when it requires the capabilities of the MRF to provide the required service, it will act as a proxy and forward the message through the S-CSCF to the MRFC. The MRFC will act as a SIP UA and reply to the SIP message accordingly. The AS maintains itself at the signalling path throughout the session. The user plane is formed directly between the terminal equipment and the MRFP. This way AS is able to provide a service that for example plays an announcement to the subscriber. This functionality has been described in the figure 14 [1].

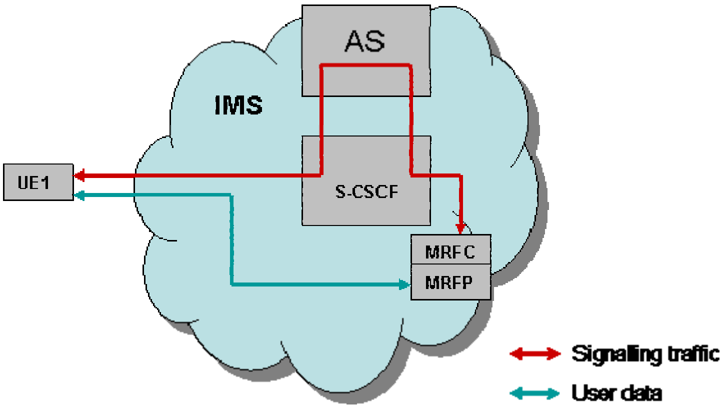


**Figure 14. The MRF functionality**

### 4.5.5. IMS interworking with CS networks

When the S-CSCF analyses the destination address of the SIP INVITE request and notices that it points to the CS network, the S-CSCF will forward the message to the BGCF. In practice this is achieved by setting in the SIP INVITE request the SIP URI to contain the terminal phone number and the value of the parameter user set as phone. The BGCF will resolve where the breakout to the CS network will occur, in other words to which MGCF the message will be forwarded to. The BGCF may choose whether it will keep itself on the signalling path of the session or not. The MGCF will forward the message via the SGW to the CS network, and together the MGCF and the SGW will perform the necessary protocol conversions from SIP to the SS7. The MGCF also controls the IMS-MGW regarding the setup of the user plane for the session. When the media is transferred, it will be transported via the IMS-MGW that will convert the CS voice to RTP stream and vice versa. The IMS interworking with CS networks has been described in the figure 15 [1].

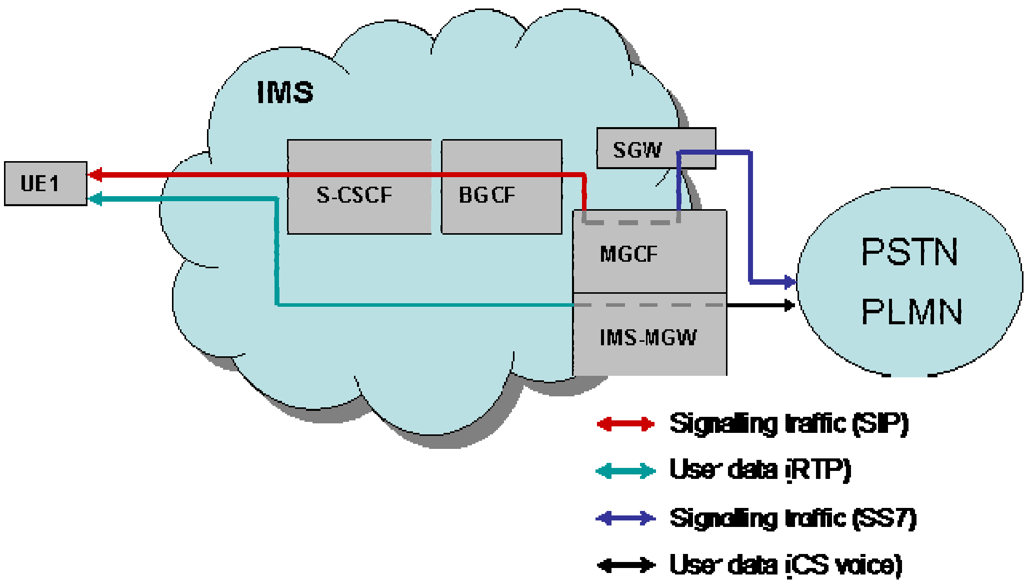


**Figure 15. The IMS interworking with CS networks**

### 4.5.6. Use of tel URI

The format of the tel URI is close to a regular telephone number. Only two things separate these two address types. Firstly a tel URI always starts with the “tel:” and secondly in a tel URI it is allowed to use separator characters. A tel URI may be used to point to a regular IMS subscriber with IMS terminal or to a subscriber in the PSTN or the PLMN. The routing in the IMS, however, is always based on the SIP URI. This means that before the message may be routed forward, the tel URI must be converted to the SIP URI. For this purpose the S-CSCF utilises the ENUM. The ENUM is a mechanism in which the telephone number of the tel URI is reversed and dots are placed after every number. Then as a domain name the “e164.arpa” is placed at the end of the string. The formed domain name is then placed into the DNS query and if there is a match, the reply from the DNS will contain the SIP URI’s of the subscriber. If there is no match in the DNS, the tel URI is pointing to the PSTN and the message is routed towards the BGCF. When using the DNS resolved SIP URI there is a problem with how to route a message to the CS terminal of the subscriber because also SIP URI’s of the IMS terminals may be attached to the tel URI. The solution to the problem is that the messages are routed to the CS terminal with SIP URI in which the user part is the telephone number and phone as a value of the parameter user [1].

### 4.5.7. IMS conferencing

The IMS conferencing functionality concerns four of the IMS functions, the terminal, Application Server (AS), Media Resource Control Function (MRFC) and Media Resource Function Processor (MRFP). An interesting issue with the IMS conferencing solution is that the interface between the AS and the MRFC is left unspecified. This means that the interface may be vendor proprietary or these two separate entities might be integrated. Together the AS and the MRFC shall form a conference focus. The conference focus is a function that creates the conference service. No extra functionality from the terminal is required for the IMS conferencing. The MRFP is responsible for the mixing of the media and the exchange of the user data with the participating terminals.

There are two different options with which an IMS conference may be created. The first is when the terminal 1 sends SIP INVITE request to the conference factory URI. The conference factory URI is previously known to the subscriber 1. No means for retrieving it automatically have been specified. The conference factory URI matching the triggering criterion in the S-CSCF will result the message to be routed towards the AS/MRCF. The AS/MRFC will then verify that the conference factory URI is allocated, authorize the user and allocate a temporary conference URI and finally allocate a conference URI for the requested conference. After receiving the original SIP INVITE the AS/MRFC will reserve the resources to the conference from the MRFP and send SIP 183 Session progress to the terminal 1 including the allocated temporary conference URI in the SIP contact header and the isfocus option tag. After this the normal resource reservation procedure is applied and when the AS/MRFC sends the final 200 OK it will include the isfocus option tag and the allocated conference URI in the contact header. At this point the IMS conference has been created. The other option for creating the IMS conference is to use a pre allocated conference URI instead of the conference factory URI.

There are two options for how the subscribers may be invited to take part in the conference. The subscriber 1 may either send the SIP REFER request to the subscriber 2 or to the conference URI. If the SIP REFER request is sent to the subscriber 2 it will contain the conference URI in the Refer-To header with method parameter set to INVITE. The subscriber 2 may join the conference by sending an

SIP INVITE request to the conference URI which is routed to AS/MRFC. When subscriber 2 receives the final SIP 200 OK from AS/MRFC the subscriber 2 has joined the conference.

If the SIP REFER request is sent to the AS/MRFC it will contain a Refer-To header including the subscriber 2 SIP or tel URI with the method parameter set to INVITE. The AS/MRFC will send the SIP 200 OK to the subscriber 1 and the SIP INVITE with the P-Asserted-Identity header containing the conference URI and the Contact header with the conference URI and the isfocus parameter to subscriber 2. When the IMS session setup is finished the subscriber 2 has joined to the conference. The creation of the IMS conference and the two different ways to invite a subscriber to the existing IMS conference have been described in the figure 12 [38].

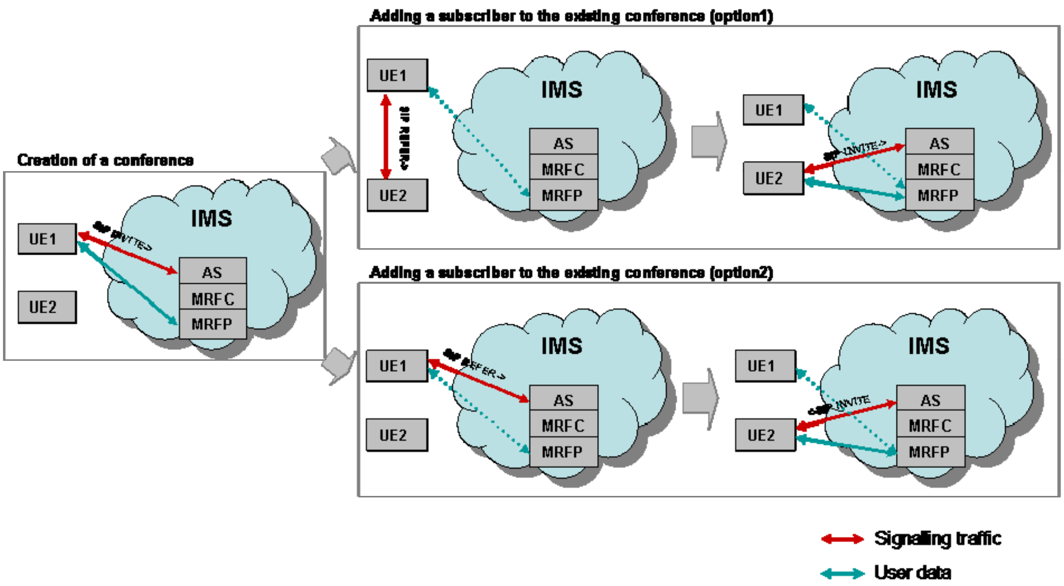


**Figure 16. The creation of the IMS conference**

# 5. Circuit Switched IMS Combinational Services

*This chapter will introduce the concept of the combinational services. It will also cover the current status and the different phases of the combinational services standardization. The architecture and the functionality are also described. Finally this chapter will address the problems and the future of the combinational services.*

## 5.1. Overview

The idea of the Circuit Switched IMS Combinational Services (CSICS) is to combine simultaneously used CS call and IMS based services in such a manner that the subscriber will perceive them as single service. A simple example of the combinational services is the combined video and speech session in which the voice part is a regular circuit switched voice call and the video part is provided as an IMS based service. These combinational services are meant to be offered via both GSM/EDGE Radio Access Network (GERAN) and UMTS Terrestrial Radio Access Network (UTRAN) [3]. The concept of the combinational services is described in the figure 17 [39].

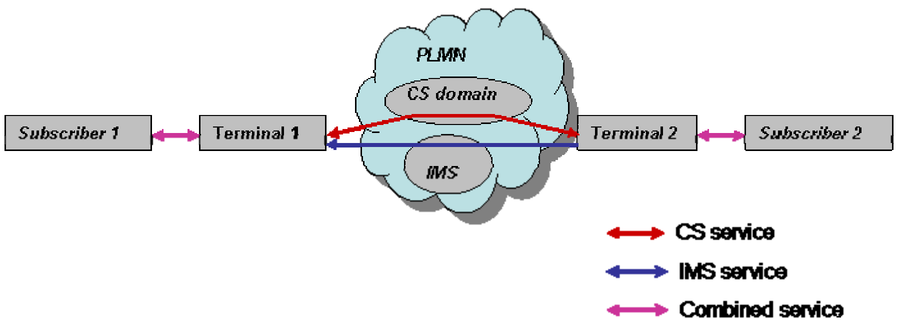


**Figure 17. The concept of the combinational services**

The CSICS is in practice a set of enriched call services. The transferring of the voice part as a CS call has two benefits. First the level of service of the voice part will be equal to regular CS call. The second is that the transferring of the voice data via CS domain releases capacity from the packet data connection. This is a benefit especially with the low bandwidth radio access networks such as GERAN because the GERAN is not able to provide acceptable QoS for the packet based transfer of real-time data [2]. The downside of the CSICS is obvious, both the terminal and the network are required to support the simultaneous usage of the CS and the PS connection.

This leads to the situation where the network might provide the support for the CSICS functionality but the terminal does not or vice versa. That is why a mechanism is required for verification that the CSICS service is supported by both the network and the terminal. Whether the two data streams will end to the same destination if they are routed via two different networks is an issue that also should be studied.

## 5.2. Standardization

The desired CSISC functionality was noted in the early stage of the standardization process to be quite extensive and require a lot of work to be fully accomplished. That is why the task was divided into two phases [40]. The phase 1 of the CSICS contains only the basic functionality and is included within the scope of the 3GPP release 7. It was created to fulfil the most urgent market needs. In practice the CSICS phase 1 includes the means for the network and the terminal capability exchange, the addition of the CS call to the existing IMS session and the addition of the IMS session to the existing CS call [3]. The CSICS phase 2 will contain the functionality left outside of the scope of the CSICS phase 1 including the interworking with VoIP terminals and the IMS control over the CS bearers [40]. The specifications available today are concerning mainly the CSICS phase 1 and even these are at the moment unfinished. Some information about the CSICS phase 2 features may be found from the feasibility studies and the 3GPP WG meeting notes.

The whole CSICS concept is included for the first time in the still unfinished 3GPP release 7. At the moment the 3GPP has produced five specifications regarding the CSICS. Three of them are pre-study like TR’s containing views of the different vendors about the desired functionality and implementation. The ongoing standardization process itself is focusing on the stages 2 and 3. This means that the included services and the architecture are specified but the detailed functionality is currently not specified. However, the TR 24.879 for the stage 3 specification is available. It gives an overview on what may be expected from the actual specification. Also it should be noted that at the moment the 3GPP release 7 is not frozen which means that the specifications available at the moment may change in the future. These changes may be anticipated by studying of the 3GPP WG meeting reports. The current status of the CSICS standardization is described in the figure 18 [5].

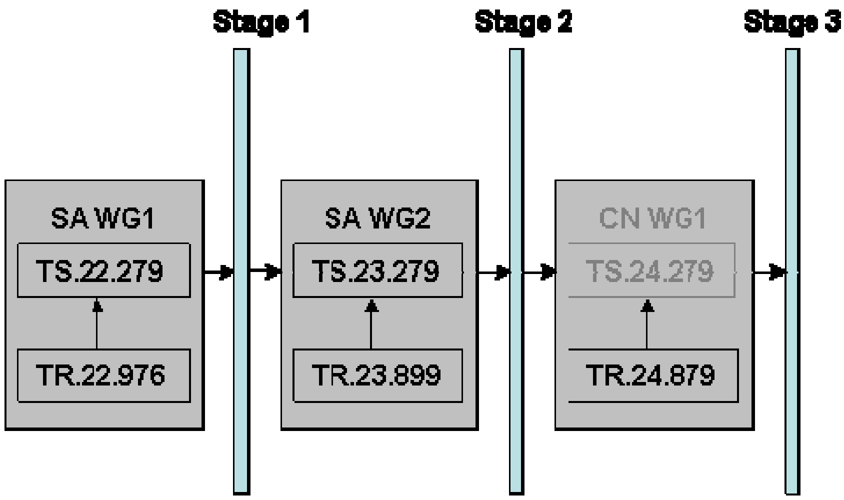


**Figure 18. The CSICS standardization status**

## 5.3. Architecture

The architecture defined for the CSICS phase 1 is very simple. It is indented to function only inside of the PLMN or between two PLMN's. The service is offered via GERAN and UTRAN, and the functionality is implemented to the terminals. The CSICS terminal has three new requirements that it must meet. In addition to these new requirements it of course has to be capable of establishing both the IMS session and the CS call separately.

First of the new requirements is the support for the simultaneous CS and PS connections. This feature in the GERAN capable terminals is called the Dual Transfer Mode (DTM) and in the UTRAN capable terminals the multiRAB. The second requirement is the capability exchange between the terminals and the third is the capability to present the CS call and the IMS session as a single service to the subscriber. The second requirement requires that the User to User Signalling 1 (UUS1) supplementary service is supported by the terminal. The third requirement is fulfilled by the application software in the terminal [39].

For the core network packet switched domain no extra functionality is required but the core network CS domain is required to provide the support for the UUS1 supplementary service. The IMS will need only to provide the session control and the support for the UE capability exchange mechanism. The AS has no part in the solution for enabling the CSICS, although it may be utilised to work in proxy mode to provide service-based charging. The entities of the CSICS phase 1 architecture are described in the figure 19 [39].

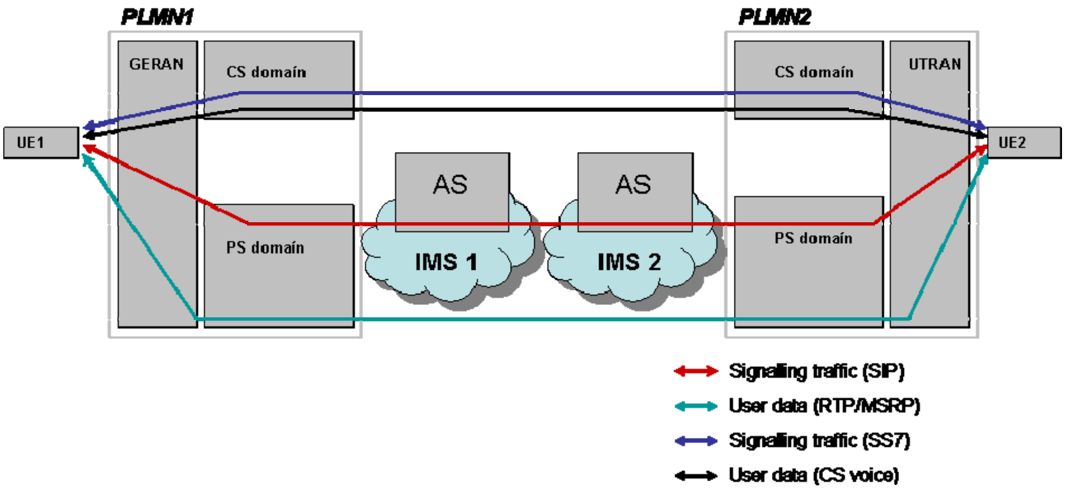


**Figure 19. The CSICS phase 1 architecture**

## 5.4. Functionality

From network point of view the CSICS does not to affect the existing services of the PLMN and requires no special provisioning. The capability to establish IMS sessions and CS calls is enough. From the subscriber point of view the CSICS has two options with the way it is specified to work [39]. First option is that the IMS session is added to the existing CS call. The second option is that the CS call is added to the existing IMS session. If for some reason another of the CSICS components must be released, the CS call has the higher priority and it will be sustained if possible [40].

### 5.4.1. Addition of an IMS session to the existing CS call

When the subscriber 1 initiates a regular CS call to subscriber 2, the information about the capabilities of the current radio network are exchanged between the subscribers utilising the UUS1 supplementary service. The exchanged information includes the ME identifier which uniquely identifies the terminal used by the subscriber. The support of radio network does not guarantee that the CSICS could be used because some of the terminals are missing the support for the CSICS. If the radio network of the remote party supports the CSICS and the terminal of the subscriber also supports the CSICS, the subscriber will initiate the IMS registration procedure. After the subscribers are registered to the IMS, the terminal capabilities are exchanged over the IMS domain with the SIP OPTIONS method. If the SIP URI of the remote party is not available to the subscriber, the tel URI derived from the remote party MSISDN is used. The exchanged information within the SIP OPTIONS method may contain information about IMS media types supported, MSISDN and preferred SIP URI, ME Identifier, CS video and voice telephony capability, MMS version supported and other supported IMS based services [39].

Because it makes sense to reduce the signalling traffic to the minimum possible the terminal is required to store the exchanged terminal capability information. This requirement causes a problem if the subscriber for example changes the terminal but still uses the same smart card and therefore the same MSISDN and the same public user ID with the different terminal. In this case the stored information about other subscriber’s terminal may not be accurate. This is resolved with the ME identifier unique to the terminal. The information about the current radio environment is not stored because of the often changing nature of this information [39].

The ME identifier exchanged with radio network capability exchange verifies whether terminal capabilities are changed. At this point all necessary information is gathered and if both the network and the terminals support the necessary functionality, the terminals offer to the subscribers the possibility to add the IMS session to the existing CS call. If the IMS session is initiated, it is done as any other IMS session. The terminals, however, will present the two exchanged data streams trough a single user interface. The addition of an IMS session to an existing CS call has also been illustrated in figure 20 [39].

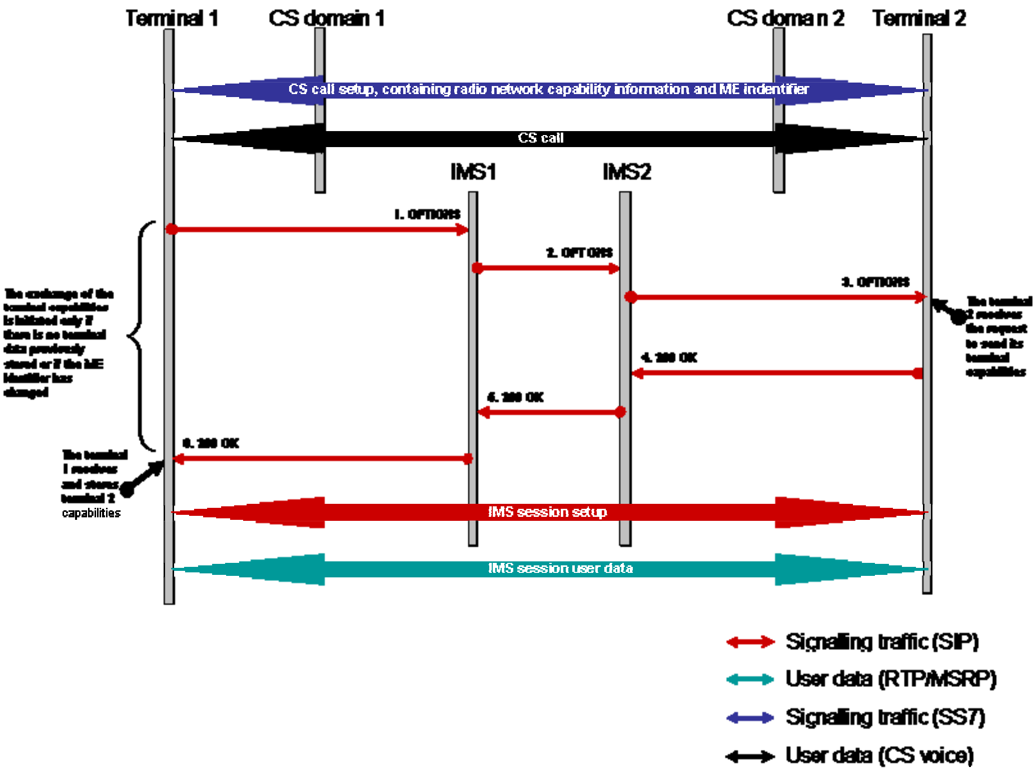


**Figure 20. The addition of an IMS session to the existing CS call**

### 5.4.2. Addition of the CS call to an existing IMS session

When subscriber 1 initiates an IMS session with subscriber 2 using a CSICS capable terminal, the capabilities of the network and the terminal including the MSISDN's of the subscribers are exchanged within the SIP INVITE method. After the IMS session is established, the terminals will offer to the subscribers the possibility to add a CS call to the existing IMS session if the use of the CSICS is possible.

When the terminal 1 initiates the additional CS call, the terminal 2 will recognise the incoming CS call as being part of combinational service with the existing IMS session based on the Caller Line ID (CLI). The CS call part of the combinational service is initiated as a regular CS call. The terminals will now present the two data streams as a single service to the subscribers. The addition of the CS call to the existing IMS session has also been describer in figure 21 [39].

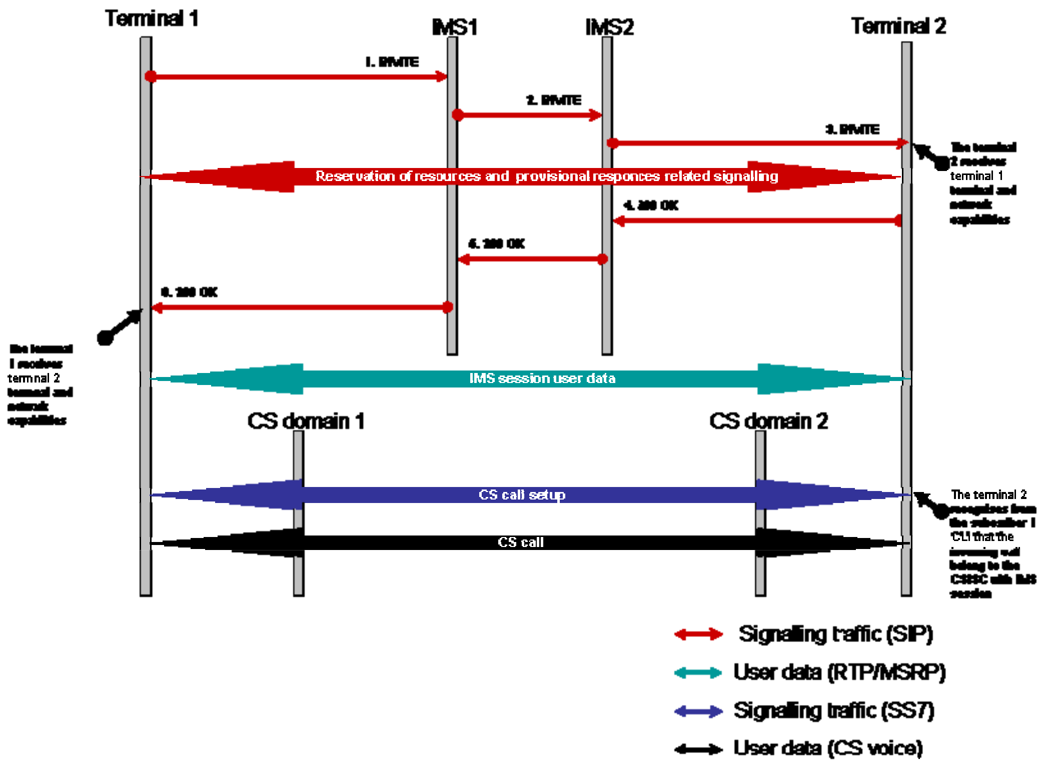


**Figure 21. The addition of the CS call to the existing IMS session**

### 5.4.3. Recognized problems

The first problem with the functionality described comes with the exchange of the current radio environment information when adding an IMS session to the existing CS call. This requires that the UUS1 supplementary service is supported by the terminals. The UUS1 however is not supported by some of the currently available UMTS terminals implemented according to the 3GPP release 5 specifications [39]. These terminals do support the multiRAB functionality and are therefore otherwise capable to use the combinational services. In practice this means that the absence of the radio network information within the CS call setup signalling does not guarantee that use of the combinational services is not possible. If the utilization of the UUS1 supplementary service is not possible, the terminal is not able to receive the remote party’s ME identifier. This means that the terminal has no means to verify if the terminal capabilities have been changed.

More problems arise when investigating the CSICS support for the supplementary services. Because the CSICS is designed to co-exist with the supplementary services the topic has been covered in the specifications. The use of the CLIR supplementary service will in practice prevent the use of combinational services.

This is because when the called party adds an IMS session to the existing CS call the routing is done based on the URI derived from the CLI of the remote party. The result is the same when the calling party adds a CS call to the existing IMS session because in this case the called party recognises the caller of the incoming CS call based on the CLI of the calling party. With the COLR supplementary service a similar problem occurs when the called party of the CS call initiates an IMS session and the calling party's terminal uses the Connected Line Identification (COL) correlate an incoming SIP INVITE with the CS call [39].

Next problem arises with the call transfers which all cause problems to the use of the CSICS. This is because the CS call transfer rules do not apply to the IMS. This leads to the situation in which the IMS session initiation and the CS call setup are routed to different destinations. The use of the ECT supplementary service is also a problem. If the terminal A has an active CSICS session with the terminals B and C and then the terminal A invokes the ECT supplementary service. the subscribers B and C are not able to invoke CSISC because the CLI has not been exchanged between them [39].

The call waiting and the call hold supplementary services work similarly with the CSICS as with regular CS calls. The multi call and MPTY supplementary services have been ruled out of the scope of the currently specified service. The call barring supplementary services are functioning with the CSICS as with regular CS calls and therefore might prevent the usage of CSICS. In this case however the IMS sessions are not restricted [39].

The interworking with the CAMEL based services creates a similar problem as with some of the supplementary services if the CAMEL based service requires manipulation of the MSISDN's of the participants. An example of this kind of service could be the use of the short numbers. When the called number is the short number of the called party, the tel URI required to reach the caller via the IMS cannot be formed since the full e164 number would be required.

The charging with the CSICS is quite challenging. From the network point of view there are two separate services that are used independently from each other. This is because all the intelligence for the CSICS phase 1 service is located in the terminal equipment. The subscriber, however, experiences these services as one and it is possible that he would like to be charged for the use of the CSICS instead for the use of the two separate services. This means that somehow the charging information of these two separate services should also be possible to be combined. This requires heavy calculations in an external database.

## 5.5. The future of the CSICS

The ongoing standardization work in the 3GPP WG's regarding the CSICS is mainly concerned with improvements of the phase 1 functionality. The capability information exchange and the update procedures have been under discussion together with the triggering of the IMS registration [41]. Also it has been decided that interesting study about IMS based services utilising the CS bearers will be performed in the future [42]. As a summary, it could be said that no dramatic changes seem to be coming to the CSICS phase 1 solution presented in this chapter. The market needs for these types of services have been so high that the major vendors have started to develop their products before the standardization is finished. At the moment at least Nokia and Ericsson have already published their first products utilising the combinational services [43][44]. Both of these products contain the sharing of the video in addition to the traditional CS call. This type of situation may cause interworking problems with solutions of the different vendors. With these video sharing types of applications the interworking problems are tackled at the moments in the video sharing test campaign arranged by the GSM Association [17]

## 5.6. Summary

The general idea of the CSICS is to use the CS services for the voice connections and the IMS sessions for other types of connections between two terminals. These simultaneous connections between two terminals are then combined in the terminals in such a manner that the subscriber perceives them as a single connection. The standardization work of the CSICS is divided into two phases and yet unfinished. The first phase contains only the means to verify that the remote party of the existing connection is capable of utilising the CSICS and the addition of the secondary connection to the existing connection. This type of service is not possible to provide without problems. Use of many of the supplementary services in practice prevents the use of the CSICS. For example the Number Identification supplementary services and the Call Offering supplementary services will prevent the usage of the CSICS. At the moment the ongoing standardization work is concentrating on finishing the phase 1 functional specifications. From the CSICS phase 2 functionality very little information is available at the moment.

# 6. Conference using combined CS and IMS based services

*This chapter presents the approaches recognised for creation of a conference service which utilises the CSICS. These approaches are studied from the point of view of the key issues recognised when studying the standardised phase 1 CSICS in the previous chapter. In addition these approaches are evaluated with each other based on the differences noticed during the study.*

## 6.1. Overview

The benefit to be achieved with the development of the conferencing service which utilises the CSICS, is to be able to offer the mobile subscriber a conferencing service with a better service experience than is possible to be achieved with the regular IMS based conferencing service. The difference is caused by the utilisation of the CS domain services for transmission of the voice media. The IMS based services are used for the transmission of the other types of media streams. The question what this chapter is trying to answer is how the service should be implemented. At the moment the IMS based conferencing service is standardised by the 3GPP, but the standardization of the conference service utilising the CSICS is not available at the moment. This chapter presents the recognised possible approaches for the implementation. At the end of the chapter the presented approaches are also compared with each other.

When investigating the different possible approaches how the conferencing service utilising the CSICS is possible to implement the number of approaches to be studied was narrowed down to the requirement that the solutions should be possible to implement utilising the functionality specified in the 3GPP release 6 with the CSICS phase 1 functionality. The focus when studying the different approaches was in resolving the issues of the CSICS phase 1. The first of these problems is how to find out if the radio network of the receiving terminal or the receiving terminal itself is capable of supporting the CSICS. The second problem is how to establish the secondary communication between the terminals. The third problem is how the receiving terminal is able to recognise that the incoming secondary connection is meant to be combined with the primary one. The floor control and the other aspects which are not directly related to combinational services were not studied. Because the release of the session is performed as with normal CSICS session it is also left outside the study.

Four different approaches to implement the conference service utilising the CSICS were recognised. The approach 1 is to create the conferencing service for the voice media in the CS domain and create simultaneously the conferencing for the other types of media in the IMS domain. The idea is the same as with the CSICS phase 1; to create the two services independently from each other and then combine the services in the terminal. Approach 1 may be implemented with the initiation of the secondary conference service as a network feature or as a terminal feature. The approach 2 differs from the approach 1 only with the location of the conference service for non-voice media. With approach 2 this is located in one of the participating terminals. In the approach 3 there is only one conference service for all types of media located in the IMS. The mobile network CS domain services are utilised only for the delivery of the voice media. The IMS based services are utilised for the delivery of the other non-voice media streams. The approach 4 is to use some proprietary conferencing solution and utilise both the CS domain and the IMS as a transport for the media streams. The figure 22 describes the four studied approaches.

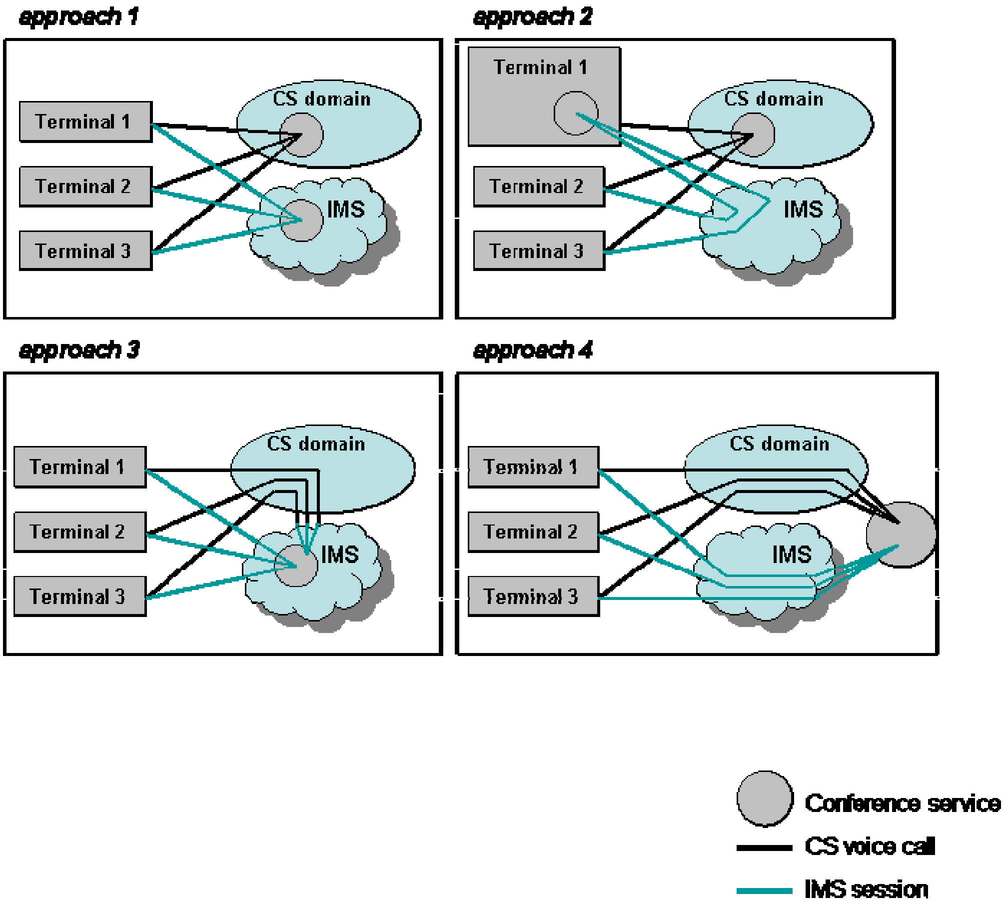


**Figure 22. The alternative approaches for the creation of the CSICS conference service to the mobile subscribers**

## 6.2. Approach 1

In approach 1 we have two separate conferences, one in the CS domain and the other in the IMS. The MPTY supplementary service is the CS domain conferencing service utilised for voice conferencing with this approach. The IMS conference service is utilised for other types of media. Four different scenarios were distinguished how the combining of the MPTY supplementary service and the IMS conference service may be implemented. In the scenario 1 the multiParty call is established first and a terminal initiated IMS conference is joined to it. In the scenario 2 the multiParty call is established first and a network initiated IMS conference is joined to it. In scenario 3 the IMS conference is established first and a terminal initiated multiParty call is joined to it. In the scenario 4 the IMS conference is established first and a network initiated multiParty call is joined to it.

### 6.2.1. Scenario 1

For a multiParty call to be established the served subscriber needs to establish normal voice calls with two other subscribers. Then with one call on hold and the other call in the active state, the served subscriber may choose to form the multiparty call between the three subscribers. In the figure 17 these events are numbered from 1 to 5. After setting the active multiParty call on hold, the served subscriber may initiate or receive calls with the subscribers which are not connected to the active multiParty call. When a new call is established the served subscriber may add the new participant to the existing multiParty call. This way the served subscriber is able to create a voice conference between a maximum of five participants [27].

Since all the subscribers are joined to the multiparty call by first initiating a normal CS call, the same mechanisms as with the normal CSICS session establishment are available for exchanging the required information. This means that when the active multiParty call is established, the served subscriber is aware of the radio network capabilities, ME identifiers and MSISDN’s of the other participants [39]. At the same time the other participants are aware only of the capabilities of the served subscriber. Because the CS call setup contained the UUS1 signalling, the terminals with CSICS capability initiate IMS registration at this point. If some of the participants would be using the standard CSICS phase 1 client they would also initiate the terminal capability exchange with the served subscriber. This is the reason why the standard client is not applicable with the approach 1.

Because the only participant in the active multiParty call which is aware of all the necessary information required for establishment of the required IMS conference is the served subscriber, it must be the served subscriber that initiates the IMS conference. However, before this the served subscriber must verify that the other participant’s terminals are CSISC capable. In the figure 23 these events are numbered from 6 and 7. If the other participant’s terminals are CSISC capable, the served subscriber creates an IMS conference by sending the SIP INVITE request to the AS/MRFC with the conference factory URI [38]. In the figure 23 these events are numbered from 8 and 9.

When the conference is allocated, the other subscriber must be invited to join in it. To achieve this we have two options. The first option is that the served subscriber requests the other participants to join to the IMS conference. The second option is that the served subscriber requests the AS/MRFC to invite the other participants to join to the IMS conference. With the first option the other participant is requested to join to the existing IMS conference by forming the SIP URI or the tel URI from participants MSISDN's and sending the SIP REFER messages to the participant utilising the generated URI. This message contains the conference URI in the Refer-To header. After receiving the message the participant will send the SIP INVITE to the AS/MRFC containing the conference URI in the Refer-To header. This results that the AS/MRFC will join the participant to the existing IMS conference. In the figure 23 these events have been numbered from 10 to 13. The new participant may use the sender information of the SIP REFER message to resolve that the IMS session created should be combined with existing multiParty call.

With the second option the other participant is requested to join to the IMS conference by sending the SIP REFER to the AS/MRFC. The SIP REFER message contains the URI of the requested participant in the Refer-To header. After receiving the message the AS/MRFC generates a SIP INVITE including the conference URI and sends it to the participant requested to join to the IMS conference. In the figure 23 these events have been numbered from 14 to 17. The problem with this option is that requested participant must be able to understand to combine the IMS session with the existing multiParty call. There is a problem because the SIP INVITE is coming from the AS/MRFC instead of the served subscriber which initiated the original CS call. This may be resolved with forming the conference URI in a manner that it contains the MSISDN of the served subscriber.

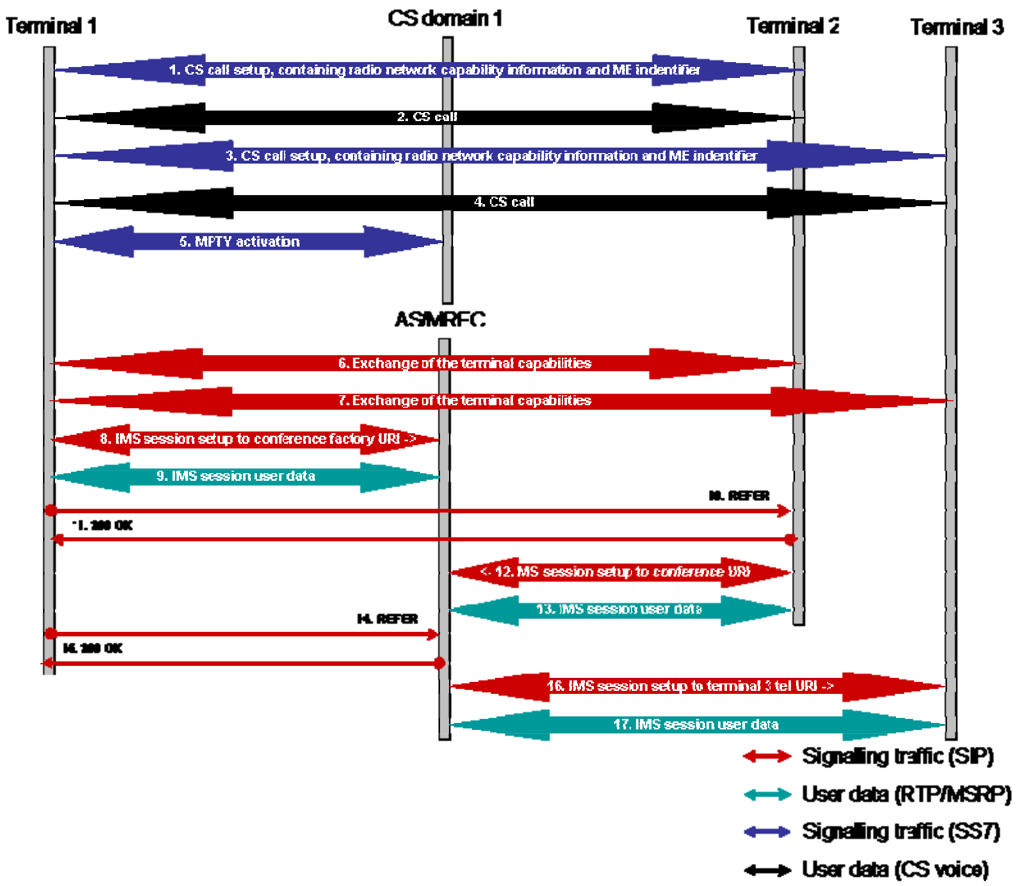


**Figure 23. Terminal initiated IMS conference joined with the existing multiParty call**

### 6.2.2. Scenario 2

In scenario 2 the multiParty call is established first in a similar manner as in the scenario 1. In the figure 24 these events have been numbered from 1 to 4. The CSICS phase 1 standard client may not be used with scenario 2 because they would initiate the capability exchange immediately after the establishment of the CS call. How the scenario 2 differs from the scenario 1 is with how the IMS conference is established. When the MPTY supplementary service is invoked by the served subscriber, the service logic in the CAMEL is triggered by the gsmSSF connecting to the serving MSS. In this scenario the AS/MRFC must have the capabilities of the IM-SSF to be able to interact with the CAMEL service logic. The CAMEL service logic informs the AS/MRFC about the MSISDN's of the participants of the active multiParty call. These events have been numbered in the figure 24 from 5 to 7.

When aware of the MSISDN's of the participants, the AS/MRFC is able to exchange the capabilities of the participating terminals by using the SIP OPTIONS method [39]. In the figure 24 these events have been numbered from 8 to 10. For this to succeed the participants should be registered in the IMS at this point. This is accomplished by the use of the UUS supplementary service within the original CS call initiation and programming the client applications to initiate the registration to the IMS when UUS signalling is noticed.

If the participants are capable of supporting the CSICS functionality the AS/MRFC will generate an IMS conference and invite the participants to join to it. In the figure 24 these events have been numbered from 11 to 16. Both the SIP INVITE and the SIP OPTIONS messages are sent to the participants using the SIP URI or the tel URI generated from the MSISDN of the participant. The problem still remaining is with how the participants are able to recognize that the incoming IMS session is meant to be combined with the existing CS call. This is possible to be resolved with by forming the conference URI in a manner that it contains the MSISDN of the served subscriber. The entire scenario 2 is described in the figure 24.

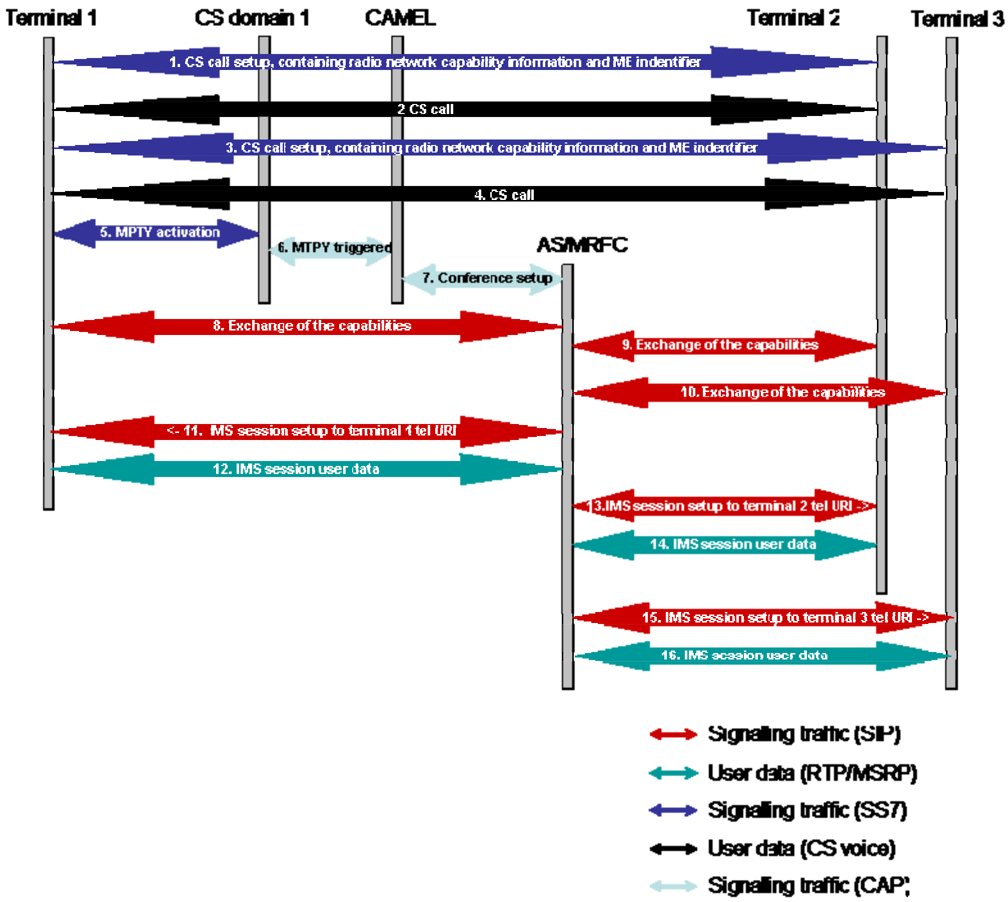


**Figure 24. The addition of the IMS conference to the existing multiParty call using the CAMEL to trigger the event**

### 6.2.3. Scenario 3

In the scenario 3, after the IMS conference is allocated, the terminal responsible for the initiation of the IMS conference also invites the other participants to join to the conference and establishes the CS calls between the participants. The IMS conference is established by a terminal sending the SIP INVITE message to the AS/MRFC using the conference factory URI [38]. As a result the terminal will be joined to the allocated IMS conference. In the figure 25 these events have been numbered as 1 and 2. The IMS conference is also possible to be pre-allocated with other unspecified means. In this case, instead of the conference factory URI, the conference URI may be used.

When the IMS conference is allocated, the subscriber responsible for the initiation is able to invite other subscribers to take part in the IMS conference by sending the SIP REFER message to the other participants with the conference URI as a value of the Refer-To header or by sending the SIP REFER message to the AS/MRFC with the other participant SIP or tel URI as a value of the Refer-To header. If the SIP REFER message was sent to the participant, it is the participant that initiates the SIP session with the AS/MRFC. If the SIP REFER message was sent to the AS/MRFC, it is the AS/MRFC that initiates the SIP session with the participant.

After inviting the other participant to join in the conference, the terminal of the served subscriber verifies that the participant is capable of establishing a CSICS connection. The current radio network capabilities and the terminal capabilities of the participants must be verified with a separate SIP OPTIONS method because the exchange of this information is not specified within the SIP REFER method. If the participant is capable of supporting the CSICS, the terminal of the served subscriber initiates a CS call with the participant. When the amount of the participants has increased to more than two, the multiParty call between participants will be established by the terminal of the subscriber responsible for the initiation of the IMS conference. In the figure 25 these events have been numbered from 3 to 17.

If the SIP REFER message was sent directly to the participant, the sender information of the message and the CLI of the incoming CS call may be used to recognise that the existing IMS session should be combined with the incoming CS call. This is not possible if the SIP REFER was sent to the AS/MRFC because the SIP INVITE message will come to the participant from the AS/MRFC. Because the CLI of the incoming CS call indicates only the MSISDN of the calling party, the conference URI must also be allocated in a manner that it contains this MSISDN. This way the participant may use the CLI to recognise that the two connections are meant to be combined.

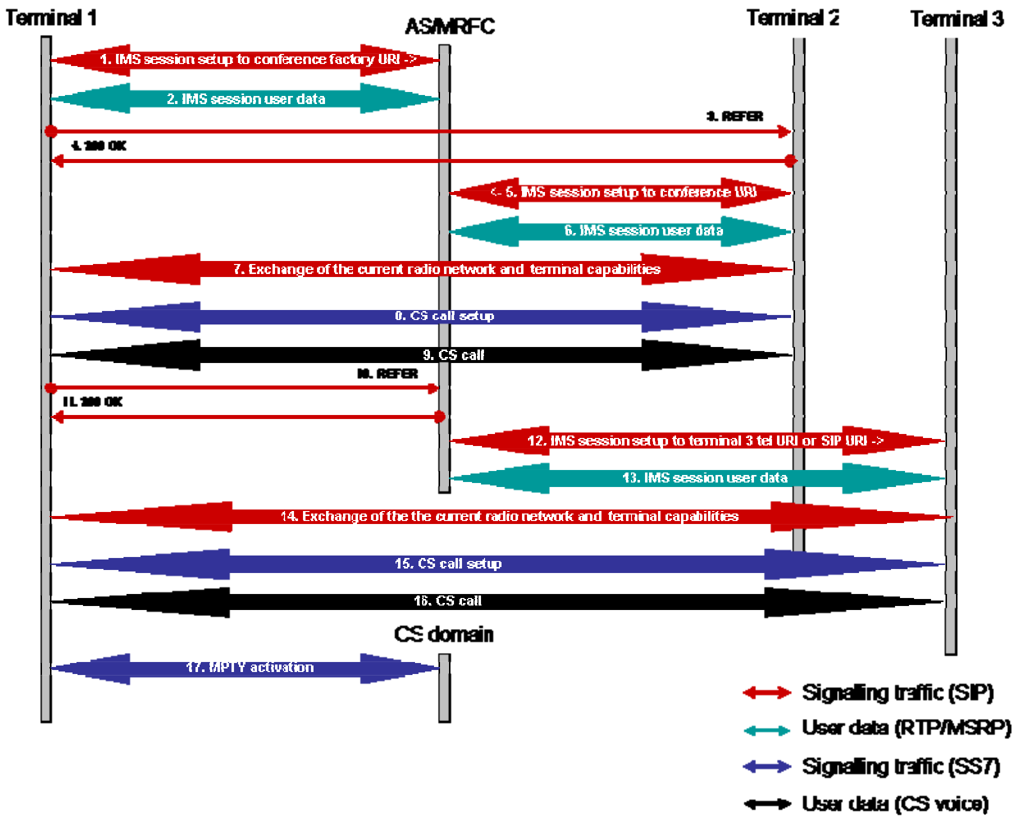


**Figure 25. The addition of the multiParty call to the existing IMS conference**

### 6.2.4. Scenario 4

In scenario 4 after the IMS conference is allocated, the CS calls between the participants are initiated as a network feature. First the IMS conference is established and the participants are invited in a similar manner as with scenario 3. In the figure 26 these events have been numbered from 1 to 4.

When a participant joins to the IMS conference, the request to establish a CS call between the participant and the subscriber responsible for the initiation of the conference will be sent to the service logic located in the CAMEL. To achieve this the AS part of the AS/MRFC must be able to act as an IM-SSF and be able to communicate with the gsmSCF. The CAMEL service logic will then initiate a CS call between the two participants. In the figure 26 these events have been numbered from 7 and 11.

The request to establish a CS call with a new participant when the number of the active participants is more than two will result a request to the CAMEL service logic to form a multiParty call between the subscribers or to add the new participant to the existing multiparty call. In the figure 26 these events have been numbered from 12 and 19.

After the IMS sessions are active, the AS/MRFC is aware of the current radio network and the terminal capabilities of all the participants because this information was exchanged within the initiation of IMS sessions [39]. The conference URI and the CLI must also in this scenario be selected in such a manner that the receiving participant is able to recognise that the two connections are meant to be combined. The last problem remains with how the CAMEL service logic will select which one of the participants of the existing IMS session will become the served subscriber of the multiParty call. This may be resolved with selecting the served subscriber to be the same participant who initiated the IMS conference.

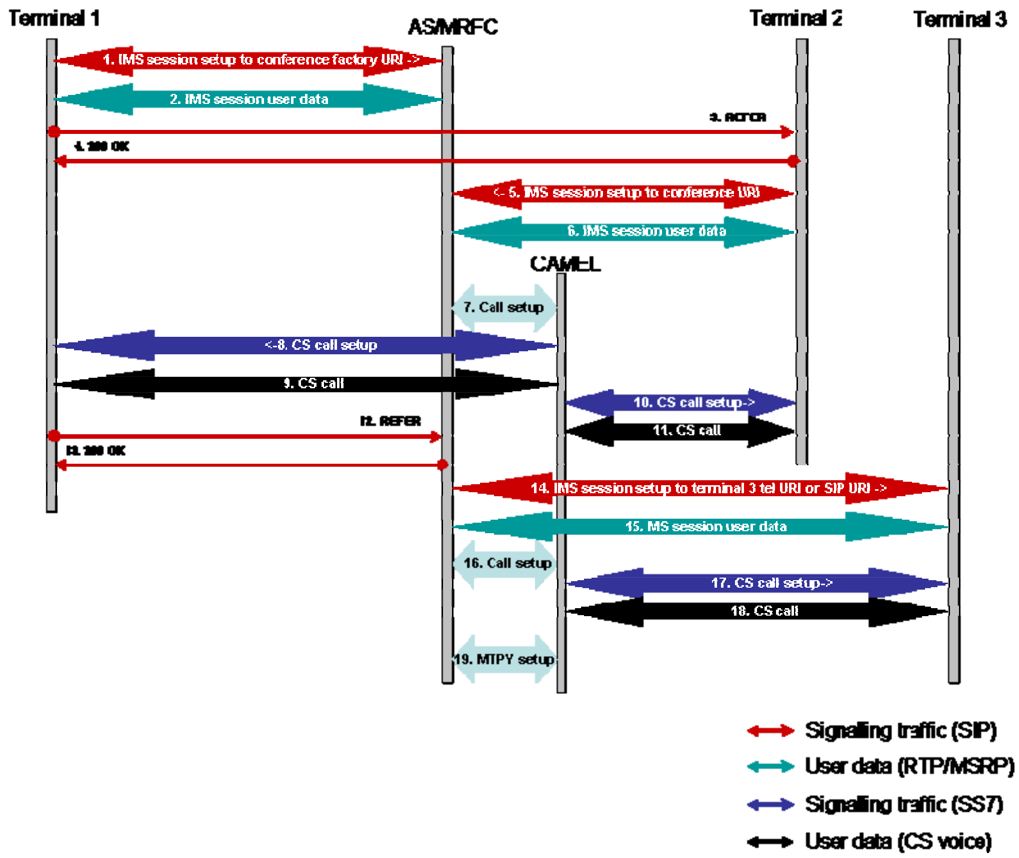


**Figure 26. The addition of the multiParty call to the existing IMS conference using CAMEL to form a multiparty call**

## 6.3. Approach 2

The approach 2 is a solution in which the conference service for voice is provided with the MPTY supplementary service and the conference service for other types of media streams is implemented in the terminal of the served subscriber of the multiParty call. The conference service implemented in the terminal of the served subscriber utilises the IMS only for the transport of the media streams. Two separate scenarios were distinguished with how the MPTY supplementary service is possible to be combined with the conference service located at the terminal of the served subscriber. In the scenario 5 the CS call between the participants is established prior to the IMS session between the participants and in the scenario 6 vice versa.

### 6.3.1. Scenario 5

In the scenario 5 the subscriber which is initiating the CSICS conference will first make a standard CS call to one of the participants. The use of the UUS1 supplementary service in the call initiation will cause the receiving party to initiate the IMS registration procedure. Then the capability exchange is performed with the SIP OPTIONS method. If both of the parties support the CSICS then an additional IMS session is established between the parties. In the figure 27 these events have been numbered from 1 and 5. Because the IMS session is established directly between the participants, the receiving terminal may use the sender's information of the SIP INVITE message to recognise that the sessions are meant to be combined.

When the third subscriber is invited to the conference, the active CSICS call is put on hold and the CSICS session with the third participant is established in the same manner as with the second participant. In the figure 27 these events have been numbered from 6 and 10. When the CSICS session is established between the two participants, the initiating participant will activate the MPTY supplementary service and form a multiParty call between the participants. Simultaneously the terminal of the served subscriber starts to perform mixing of the media for the active IMS sessions. The served subscriber is able to join a maximum of five participants to the CSICS conference with first placing the active multiParty call on hold, then initiating a CSICS session with the participant to be joined to the conference and finally adding the participant to both the multiParty call and the conference of the IMS session [27]. In this scenario at least the served subscriber must have a special type of application. The other participants are able to participate in the conference with the standard CSICS phase 1 client.

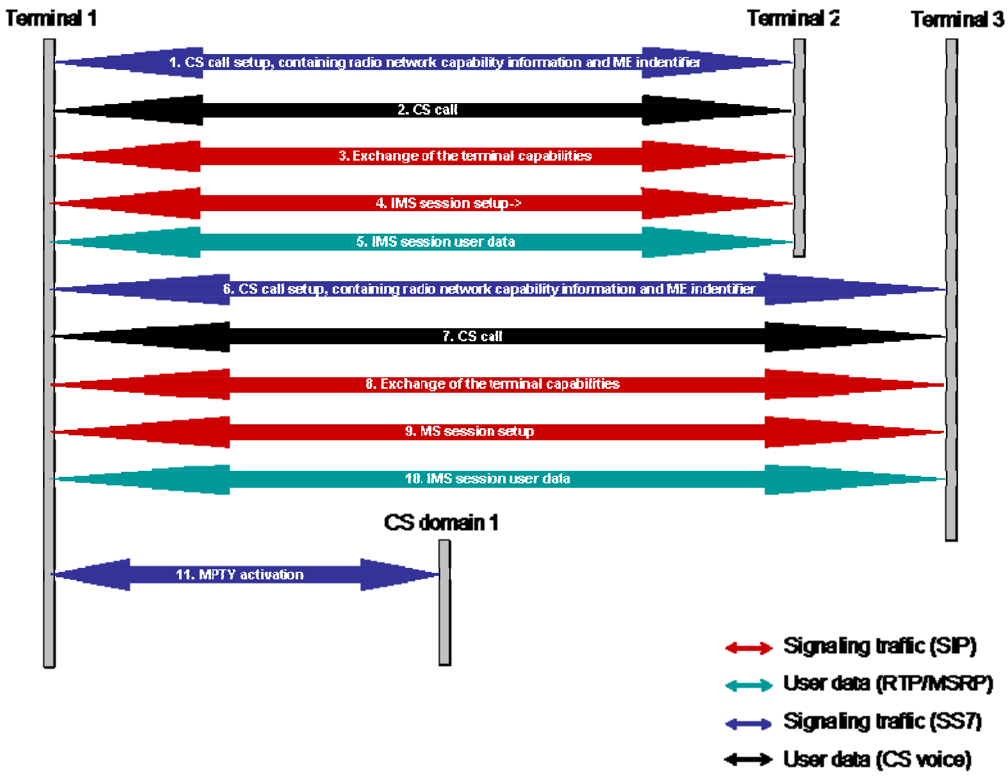


**Figure 27. The MPTY combined with the conference service implemented to the terminal when initiating the CS call first**

## 6.3.2. Scenario 6

In the scenario 6 the subscriber which is initiating the CSICS conference will first initiate an IMS session with one of the participants. The capability information is exchanged within the IMS session initiation [39]. If the remote party supports the CSICS, the calling party will initiate a CS call between the parties. The remote party is able to use the CLI of the incoming CS call to recognise that it is meant to be combined with the existing IMS session. In the figure 28 these events have been numbered from 1 and 4.

The CSICS session with the third participant is established in a similar manner after the active call is put on hold. The conference is formed when the terminal of the served subscriber invokes the MPTY supplementary service and creates the multiParty call between participants. Simultaneously the terminal of the served subscriber starts to perform the mixing of the media to the IMS sessions. This way implemented CSICS conference has a limit of maximum five participants. In the scenario 6 the special type of application is required only from the served subscriber. The other participants are able to take part to the conference with the standard CSICS phase 1 client.

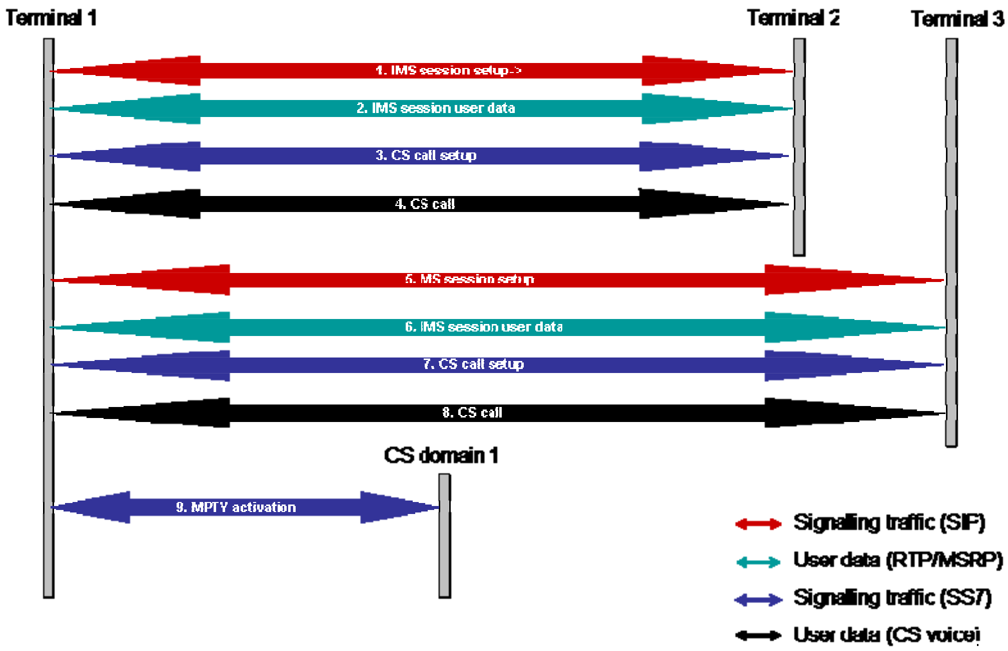


**Figure 28. The MPTY combined with the conference service implemented to the terminal when initiating the IMS session first**

## 6.4. Approach 3

The approach 3 is a solution in which the IMS conferencing service is enhanced with a capability to route the media streams of the conference via a different route to the same destination. The purpose is that the voice media is routed to the CSICS capable terminal via the CS domain. The non-voice media in turn is routed to the terminal as a normal IMS session. With the approach 3 two separate scenarios were distinguished. In scenario 7 the CS call between the terminal and IMS conference service is established prior to the IMS session. The scenario 8 follows reversed order, the IMS session is established prior to the CS call.

### 6.4.1. Scenario 7

If the CSICS utilising the IMS conference service is established with the terminal initiating a CS call with the AS/MRFC, there will be a problem with how the contents of the IMS session are negotiated. The problem is that the IMS interworking with the CS networks does not currently support the UUS supplementary service [38]. This means that the AS/MRFC does not have the information about terminal's current radio network available. The problem is possible to be resolved by the AS/MRFC always initiating an IMS session towards the terminal. It is also required that the terminal initiates the IMS registration procedure simultaneously with the initiation of the CS call. This requirement rules out the use of the standard CSICS phase 1 client. Even if the terminal is registered to the IMS, if the current radio network or the terminal does not support the CSICS, the AS/MRFC initiated IMS session will lead only to the unnecessary signalling. However it verifies if the IMS conference is meant to have only voice as media or if some additional media is meant to be used.

When the AS/MRFC initiates an IMS session back to the calling party terminal, it may use the SIP URI or tel URI derived from the CLI of the incoming CS call. This is mapped to the SIP INVITE sent by the IMS interworking function to the AS/MRFC. The temporary conference URI must be allocated in a way that it includes the conference telephone number, indicating to the terminal that the IMS session is meant to be combined with the existing CS call. In the figure 29 these events have been numbered from 1 and 6.

The conference may be joined by initiating a CS call to the conference telephone number. In principle the invitation to take part to the conference may be delivered to the terminal with either utilising the SIP REFER method or by some other means. The use of the SIP REFER method between terminals unfortunately requires that the subscriber is registered to the IMS which results in that if this method is used, the SIP REFER must be sent to the AS/MRFC. In this case the AS/MRFC would be able to initiate a CS call towards the invited terminal, but the problem encountered then is that if the terminal does not detect the use of the UUS supplementary service it does not initiate the IMS registration procedure. This is why we are left with only the option of inviting the other subscriber to take part to the conference with some external means. An example of these external means might be the Short Message Service (SMS).

After receiving the invitation, the subscriber will initiate a CS call with the conference service. After the establishment of the CS call, the AS/MRFC will initiate an IMS session towards the terminal of the subscriber. The capabilities are exchanged during the signalling and if accepted by the terminal the additional non-voice IMS session is established in addition to the existing CS call. The conference URI must be allocated in a way that it includes the conference telephone number, indicating to the terminal that the incoming IMS session is meant to be combined with the existing CS call. In the figure 29 these events have been numbered from 7 and 12.

With the scenario 7 there is a problem with several subscribers willing to create several conferences simultaneously. This would require a unique phone number for each of the conferences. This problem may be solved for example by requiring the subscriber to indicate the particular conference in which he is willing to join or by creating a numeric code. This could be transmitted from terminal to AS/MRFC for example by using Dual Tone Multi Frequency (DTMF) after the CS call has been established.

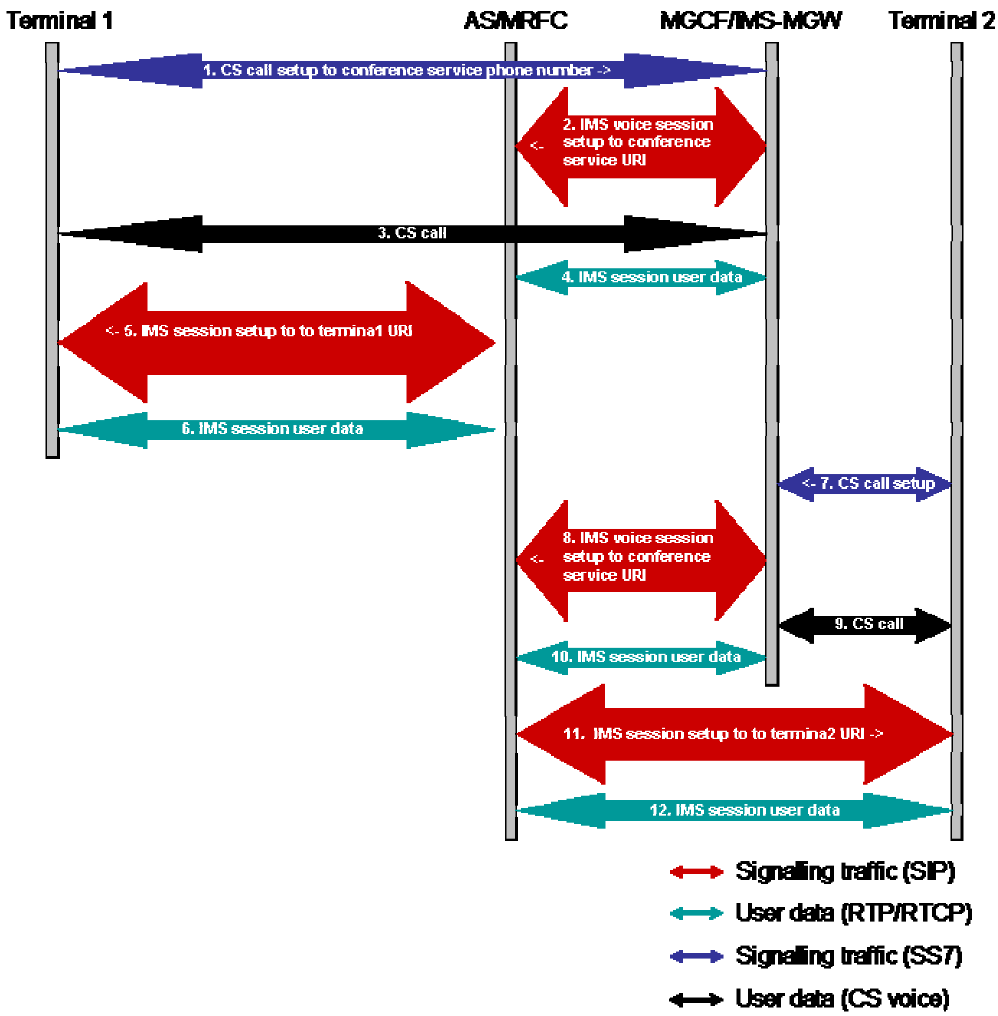


**Figure 29. The creation of the IMS conference utilising the CS domain to transport the voice media with CS call**

### 6.4.2. Scenario 8

The second scenario of the approach 3, the scenario 8, is to establish the CSICS utilising IMS conference with a terminal initiating an IMS session to the conference factory URI. If during the initiation of the conference, the terminal which is capable to establish the CSICS session realises that the AS/MRFC also supports the CSICS, it will establish the IMS session without the voice media. Then the terminal initiates a CS voice call towards the AS/MRFC in addition to the established IMS session. Since the MSISDN's of the terminal and the AS/MRFC were exchanged during the session initiation, the AS/MRFC is able to recognise that the CS call and the IMS session belong to the same conference based on the CLI of the incoming CS call. In the figure 30 these events have been numbered from 1 and 6.

The participant may be invited to take part to the existing conference by utilising the SIP REFER method as with any regular IMS conference. The invited participant may join to the conference by initiating the IMS session towards the conference URI. Again the terminal recognises that that the AS/MRFC is CSICS capable. If possible, the terminal will then establish the IMS session with the AS/MRFC without the voice media and initiates a CS voice call towards the AS/MRFC based on the MSISDN's exchanged during the IMS session setup. The CLI of the incoming call are again utilised as means to verify that the two connections to the same conference. In the figure 30 these events have been numbered from 7 and 14.

If the functionality of the scenario is slightly changed also the standard CSICS phase 1 client may be used to participating to the conference. This requires that the SIP REFER method is not utilized for the invitation of the participants.

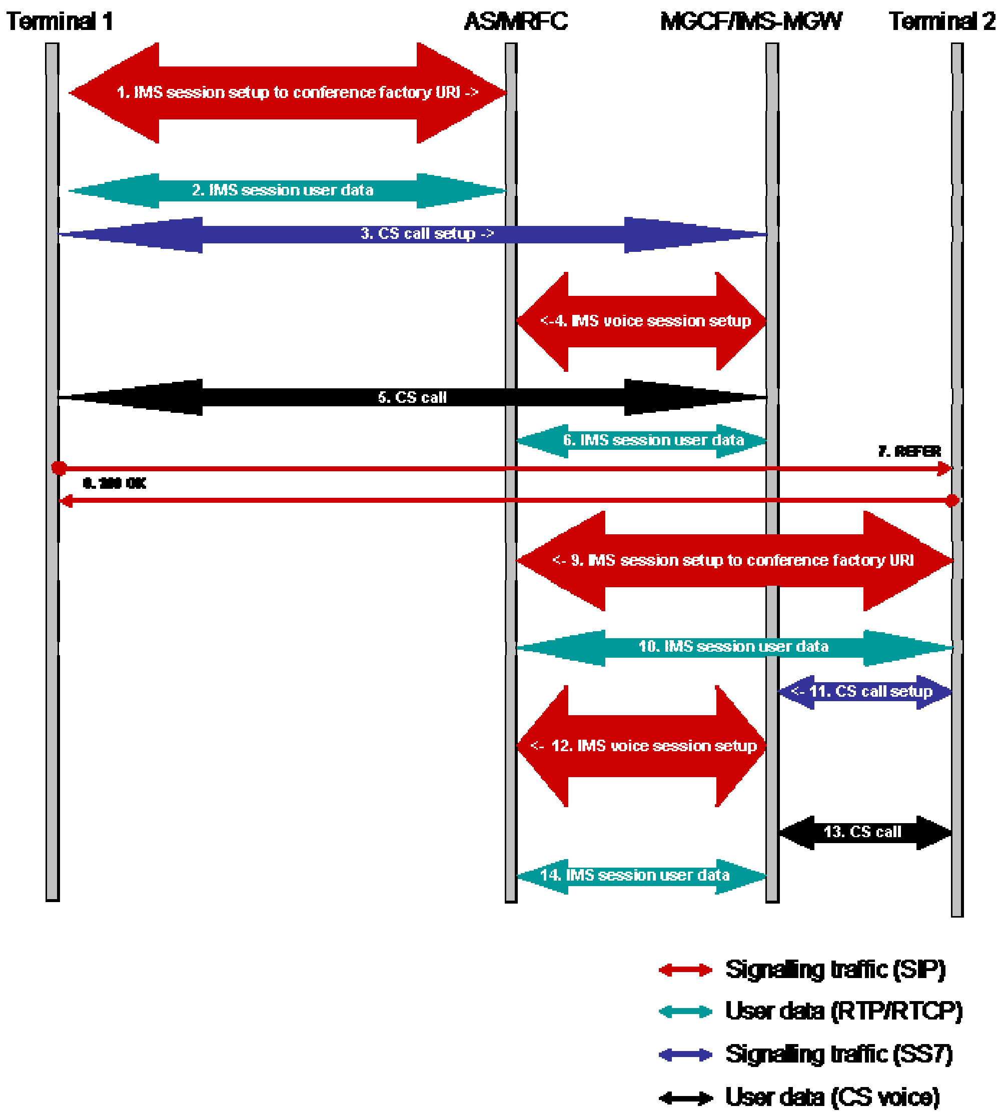


**Figure 30. The creation of the IMS conference utilising the CS domain to transport the voice media**

## 6.5. Approach 4

The approach 4 is to create a non-standardised conferencing service utilising the CS domain and IMS domain as a means for transport of the media. This way the freedom is achieved to specify the functionality to be exactly what is required. This type of conferencing service is possible to be implemented in several ways, but the additional benefit in this model is that it can also be implemented to look exactly like a CSICS capable client. This enables the use of the conferencing service with CSICS phase 1 terminal only with basic one-to-one functionality without any extra application or feature required. This type of conferencing service logic is located outside the IMS and the CS domain with having only interfaces towards both of the domains.

When the conferencing service is specified to look to the terminal exactly as any other CSICS capable client, the solution to use SIP REFER method for invitation of other participants to join the conference is not available. Also the conference must be pre-allocated via some external user interface. Then the participants are able to join to the existing conference with simply forming a CSICS session with the conference service. The signalling with this approach will be according to the figures 14 and 15.

## 6.6. Comparison of different approaches

### 6.6.1. The criteria for the comparison

The approaches presented have different characteristics and to be able to perform a comparison, some criteria's on which the evaluation is based on must be stated. In the beginning of this chapter we stated the requirement that the solution should not require other functionalities than what is specified with the CSICS phase 1 and in the 3GPP release 6. All approaches presented fulfil this requirement so to establish differences between the solutions other more specific functionality evaluation criteria's must be stated. To find useable criteria we list the differences noticed while studying different approaches.

The first difference between the approaches is the ability to be used by the standard CSICS phase 1 client. When evaluating the approaches we will consider the support for the standard CSICS client functionality as a significant benefit, but for certain type of applications this might not be an important feature. The approaches also differ with the number of conferences required to provide the CSICS conference service. If the CSICS conference service is created with combining two separate conference services, the charging will get more complicated. All solutions need the terminals to have the possibility to initiate the CS calls and the IMS session and the UUS1 supplementary service must be provisioned. The approaches differ in a manner that some approaches require also that some additional services are provisioned. There are also differences in how the services based on the different approaches behave in a problem situation and with the way they use resources of the radio interface. Some approaches also have a limitation with the number of participants. The possibility to take part to the conference with the pure IMS conference client or with client capable only to the CS voice calls is a matter that also causes differences between approaches.

From these differences the eight features in the table 9 were derived to be evaluated with each of the approaches.

| **Feature** |
|---|
| Support for the standard CSICS client |
| Support for the standard IMS conference client |
| Support for the standard CS voice client |
| Charging data may be collected from a single source |
| No additional provisioning required |
| Good behaviour in the problem situations |
| Unlimited number of participants |
| Efficient use of the radio resources |

**Table 9. The criteria's for the evaluation of the approaches**

### 6.6.2. Approach 1

With approach 1 in none of the scenarios the served subscriber of the multiparty call is able to use a standard CSICS client. The charging requires combining the charging records of the multiParty call and the IMS conference. Also the provisioning of MPTY supplementary service and, if CAMEL is utilised, the suitable CAMEL triggers are required. The using of the MPTY supplementary service for the voice audio conference causes the problem in case when the served subscriber loses the radio network coverage or runs out of battery [27]. In this case the whole multiParty call is released. This is why the approach 1 is stated to have poor behaviour in the problem situations. All the presented scenarios under the approach 1 have the limitation of maximum five participants per conference. This is due to the limitation of participants in the multiParty call [27]. The use of the radio network resources with the approach 1 is efficient since only the minimum amount of data is transmitted over the radio interface. The standard IMS conference service client may not take part in conferences, but the participation with an only CS voice capable client is possible.

### 6.6.3. Approach 2

The approach 2 requires the served subscriber to have a special type of client since the conferencing service of the non-voice media streams is executed in the terminal of the served subscriber. If the operator is willing to offer lower tariff for the combinational services compared to the combined tariff of the CS and PS services utilised for creation of the combinational service, the charging data should be collected from both the CS domain and the IMS. The approach 2 also requires that the MPTY supplementary service is provisioned. In the problem situations also the approach 2 has a poor behaviour. If the served subscriber loses the radio network coverage or the battery runs out, both of the conferences are released. The number of participants is limited to five as with the approach 1. The main disadvantage with the approach 2 is the excessive use of the radio network resources. The execution of the non-voice conference in the terminal causes increased load to the radio interface which may be a big problem, especially when utilising the GERAN. It is not possible for the standard IMS conference service client to take part to the conferences based on approach 2. The CS voice capable client is able to participate to the multiParty call.

### 6.6.4. Approach 3

With the approach 3 the CSICS client is capable of participating to the conference. This requires that the invitation to the conference must be performed with means not included to the standard client. The problem with the scenario 5 is the same as with some of the UMTS terminals, the UUS supplementary service is not supported [38]. The proper functioning of the standard CSICS client with the service is possible to be ensured by specifying that the signalling should be according to scenario 6 without the SIP REFER method. The charging of the conference service based on the approach 3 may be based solely on the charging records collected from the AS/MRFC. This is because the AS/MRFC contains the whole service logic. No additional services are required to be provisioned. If the participant will lose the radio network coverage or will run out of battery the conference service of the other participants is not affected. The number of participants is unlimited. Since in the approach 3 the support of the CSICS is only an addition to the standard IMS conference service, the non-CSICS capable IMS conference client is able to use the service. Also the option to support the participation of the client capable only to the CS voice calls to the conference is possible.

### 6.6.5. Approach 4

Since the functionality of this type of conference service is not restricted to the standardised solutions, it may be implemented to provide any kind of desired behaviour. The approach 4 has characteristics very close to the approach 3. The only actual advantage over approach 3 is the support for the UUS supplementary service.

## 6.7. Summary

Four different approaches to create the conference service utilising the CSICS were recognised. The first was MPTY supplementary service combined with the IMS conference service, the second was the MPTY supplementary service combined with the IMS conference located at the terminal of the served subscriber, the third was the IMS conference utilising the CS domain for voice transport and the fourth was the proprietary solution. The possible functionalities of these solutions were studied from the perspective of the problems of the CSICS. These included the means for exchanging current radio network and the terminal capability information and identifying the secondary connection as a connection meant to be combined with the primary one. With all presented solutions these problems may be solved and therefore they were recognised as to be possible ways to implement the service. The comparison of the approaches was based on the differences noticed during the study of the approaches. As a summary from the comparison made, the table 10 is presented.

| **Feature** | **Approach** | | | |
|---|---|---|---|---|
| | **1** | **2** | **3** | **4** |
| Support for the standard CSICS client | No | No | Yes | Yes |
| Support for the standard IMS conference client | No | No | Yes | Yes |
| Support for the standard CS voice client | Yes | Yes | Yes | Yes |
| Charging data may be collected from a single source | No | No | Yes | Yes |
| No additional provisioning required | No | No | Yes | Yes |
| Good behaviour in the problem situations | No | No | Yes | Yes |
| Unlimited number of participants | No | No | Yes | Yes |
| Efficient use of the radio resources | Yes | No | Yes | Yes |

**Table 10. Summary of the comparison of the approaches**

# 7. Conclusions and future work

*In this chapter we presented the summary of the thesis with the conclusions. In addition the topics for further research are suggested*.

## 7.1. Summary of the thesis

This thesis had two objectives. First was to study the current situation of the standardization of CSICS and the functionality the 3GPP specifications contain at the moment. The second objective was to study how the specified functionality could be utilised for providing a combinational conferencing service.

The CSICS is a service which utilises both services provided by the PLMN CS domain and the IMS. At the moment the standardization of CSICS is divided into two separate phases. Both of these phases are included in the 3GPP release 7 which is yet to be frozen. The standardization process of the 3GPP itself is divided into three stages. The first stage contains the requirements, the second stage contains the architecture and the third stage contains the detailed protocol level functionality [5]. Regarding the CSICS phase 1 this process is at second stage. This means that in practice the architecture of the service is specified but actual mechanism is still under development. The first phase of the CSICS contains only the basic functionality enabling the two terminals to establish a simultaneous CS call and an IMS session between them [40]. The intelligence of the service is built in the terminals. This means that from the network point of view these two sessions are totally independent from each other. The CSICS requires from the terminal the ability to establish simultaneous CS and PS connections [39]. With the GERAN capable terminals this feature is called Dual Transfer Mode (DTM) and for UTRAN capable terminals it is known as multiRAB. Another requirement for the terminal and the network is the capability to support the UUS supplementary service.

When studying how the CSICS could be utilised for providing a conferencing service, four different approaches were recognised. The approach 1 is that two conferences are produced separately in the CS domain and in the IMS and media's of the conferences are combined in the terminal. The idea of the approach is close to the CSICS phase 1 service. The approach 1 is possible to be divided into two different categories depending on if the CAMEL was utilised for the initiation of the secondary conference or not. The approach 2 is to create two separate conferences as with approach 1, but with the difference that the conference for non-voice media's is located at one of the participating terminals. The approach 3 will add new functionality to the standardised IMS conference service enabling the utilization of the CS service as a transport for the voice media. The approach 4 is to produce the conferencing service with a proprietary solution. The service is located outside the IMS and the CS domain. This way produced service would seem to the participant's terminals as a CSICS capable terminal.

All of these approaches were recognised as to be possible to be implemented with the CSICS phase 1 functionality combined with the 3GPP release 6 functionality. As an evaluation the differences of the presented approaches noticed while studying the functionality of the different scenarios were listed. Based on the support for multiple types of clients, unlimited number of participants and efficient use of the radio resources we concluded that the approaches 3 and 4 would be the most suitable ones for the implementation of the CSICS conference service.

## 7.2. Conclusions

In my opinion there is a clear need for the combinational services from operator point of view. This is based on the fact that the terminals have access to the public internet and may be equipped with applications using the services of the public Internet. These services are very cheap to produce because if the service is located in the public Internet the customers do not expect the same level of reliability as with the services offered by the mobile operators. In addition to this, the complicated charging functions desired by the operator's increase the cost of producing the service. These types of issues cause the operator service development to be expensive and at least at the moment it seems that the customers are more interested in the price than the reliability.

The fear of the mobile operators is that instead of selling the actual services the operators will be selling only data transfer capacity with lower profits. This is the main reason why the IMS was developed, but in my opinion utilising the IMS to provide the services proved to be success on the public Internet will not be enough. Even if the operators would be able to develop a new type of appealing application it might not create large revenue if it would be possible to be implemented also in the public Internet. The concept of the new service would be copied and offered via the public Internet. The tight competition caused by this would lead to the low profits. The benefit of the CSICS is that it enables the providing of the real-time services over the GERAN and the UTRAN, which is a feature neither possible nor available to the Internet servers.

At the moment, the standardization is in quite early stage and there is a limited amount of CSICS capable terminals available. Some of these terminals unfortunately do not even support the UUS supplementary service which reduces even further the amount of the suitable terminals. Other issue is that the CSICS phase 1 is not due until 3GPP release 7 and the currently operating networks are commonly implemented according to the 3GPP releases 4 or 5. This means that the standardised CSICS phase 1 solution and large amount of the CSICS capable terminals will probably not be the reality for quite some time.

The best approaches to implement the general type of conference service utilising the CSICS while waiting the actual standards of the phase 2 were determined to be either the approach 3 or the approach 4. The approach 3 has the benefit that the interworking with the IMS conference clients is arranged as a default feature. The approach 3 will also fit to the possible policy of the operator that new services should be created utilising the IMS. Also based on the 3GPP TR's the approach 3 will likely be close to the standardised solution in the future. The downside of the approach 3 is that the interworking with the circuit switched networks does not support the UUS supplementary service at the moment.

The approach 4 on the other hand does support the UUS supplementary service, which enables the conferencing service to be completely disguised to appear as a regular CSICS phase 1 client. Unfortunately, some networks have limitations with the maximum length of the UUS message [27]. If the approach 4 was to be selected as a model to the CSICS conference service implementation, it should be verified that the CS client part of the service is located in such a network that no additional problems are confronted.

While studying the approaches it was noted that even when not applicable to general types of conferences the approach 2 might prove to be useful with some special types of applications. For example with mobile games played with a small group of players, this approach would be an easy operator independent way to create the service. This requires that the voice connection would be important part of the game and the bandwidth required for the non-voice media would be very low. If these terms are met, approach 2 could be an option to consider. The approach 2 might prove to be usable mainly to some terminal application developers.

## 7.3. Future work

The future work that remains to be done is to verify the standardised version of the CSICS phase 1 functionality when the 3GPP release 7 will be frozen and also the CSICS phase 2 which was left outside of the scope of this thesis because of the lack of material published regarding it. When this situation changes, the CSICS phase 2 might be an interesting topic to study further. Some applications might require that the two independent data streams of the CSICS are synchronised. The means that provide this type of functionality might also be worth studying. The security aspect of the CSICS was also ruled out of the scope of this thesis. The new possible security problems raised by the utilization of the CSICS could therefore also be studied. As an interesting topic from an operator point of view would be the survey of the advantages the operator has over a service provider in the public Internet could be performed.

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## Appendixes